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THE PRACTICAL IMPACT OF RECENT COMPUTER ADVANCES ON THE ANALYSIS AND DESIGN OF LARGE SCALE NETWORKS

NETWORK ANALYSIS CORPORATION

PREPARED FOR
ADVANCED RESEARCH PROJECTS AGENCY

DECEMBER 1974

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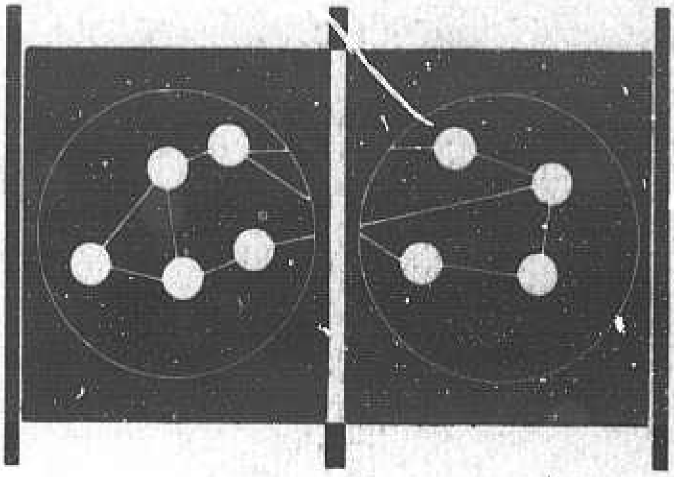
The Practical Impact of Recent Computer Advances on the Analysis and Design of Large Scale Networks

Fourth Semiannual Technical Report



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13. ABSTRACT

This report summarizes studies of local, regional, and large scale data communication network problems. Primary emphasis is placed on system issues and tradeoffs. Areas discussed are packet radio system studies, packet radio system algorithms and control, local and regional data network cost comparisons and alternatives, and integrated large scale packet switched network cost and performance. Studies of multidropped, point-to-point, and broad band cable and radio broadcast systems as well as the impact of satellites on network cost and performance are described.

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Computer networks, throughput, cost, reliability, packet switching, packet radio, broadcast radio.

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December 1974

For the Project

The Practical Impact of Recent Computer Advances on the Analysis and Design of Large Scale Networks

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SUMMARY

Technical Problem

Network Analysis Corporation's contract with the Advanced Research Projects Agency has the following objectives:

- To study the properties of packet switched computer communication networks for local, regional and large scale communications.
- To develop techniques for the analysis and design of large scale networks.
- To determine the cost/throughput/reliability characteristics of large packet-switched networks for application to Defense Department computer communication requirements.
- To apply recent computer advances, such as interactive display devices and distributed computing, to the analysis and design of large scale networks.

General Methodology

The approach to the solution of these problems has been the simultaneous

- Study of fundamental network analysis and design issues.
- Development of efficient algorithms for large scale network analysis and design.
- Development of an interactive distributed display and computational system to deal with large-scale problems.
- Application of the new analysis and design tools to study cost and performance tradeoffs for large systems.

Efforts have concentrated on the following areas:

- Packet Radio System Network Studies.
- Packet Radio System Network Algorithms and Controls.
- Local and Regional Data Network Performance and Cost Comparisons
- Integrated Large Scale Packet Switched Network Cost and Performance.
- Support Facility Development.

Technical Results

This document summarizes principal results obtained over the entire contract period with primary emphasis on the system issues and tradeoffs identified and examined. Algorithms, models, and design tools are discussed in prior semiannual reports. Accomplishments include:

- Base performance characteristics of a single station, fixed repeater location Packet Radio System were established.
- The effects on performance of a number of fundamental hardware design decisions including use of multiple or single detectors at repeaters and stations, tradeoffs between range, power and interference, single or dual rate repeaters, common versus split channel operation, and the use of omni versus directional antennas were evaluated.
- System delay, throughput and blocking under various routing alternatives, acknowledgement schemes, and repeater network organization were quantified.
- It was demonstrated that the Packet Radio System, using unoptimized operating parameters and algorithms, can provide reliable and efficient transportation of packets.
- Efficient routing algorithms were developed, simulated and tested, and based on these tests, recommended for implementation.
- Single station, multirepeater initialization, network mapping, and transmission algorithms were developed.
- Hop-by-hop and end-to-end acknowledgement schemes were developed, simulated and tested.
- Simple terminal search and local terminal control algorithms were developed and simulated.

- A repeater location optimization algorithm was developed, programmed, and tested.
- Low cost terminal access via hardware multiplexing at TIPs was demonstrated.
- The cost-effectiveness of multipoint lines for connecting low and medium speed terminals into ARPANET was shown. The use of software demultiplexing as a means of increasing the terminal handling capacity of a TIP by a factor of 10 was proven.
- The theoretical efficiency of incorporating broadcast packet radio techniques on a wideband coaxial cable local distribution network servicing a large terminal population was shown.
- Basic analysis and design algorithms for optimization of terminal processor location, topological optimization, throughput and delay analysis, and reliability analysis were completed.
- The cost-effectiveness of using satellites to increase ARPANET capacity was proven.
- The feasibility of a 1,000 IMP packet switched network using terrestrial was demonstrated.
- The cost-effectiveness of packet switching within an environment containing several thousand terminals was shown

Department of Defense Implications

The Department of Defense has vital need for highly reliable and economical communications. The results developed prove that packet switching can be used for massive DOD data communications problems. A major portion of the cost of implementing this technology will occur in providing local access to the networks. Hence, the development of local and regional communication techniques must be given high priority. The results on packet radio demonstrate that this technique can provide rapidly deployable, reliable and efficient local access networks. In addition, the initial results on the use of domestic satellites indicates that substantial savings can be achieved by their use in large scale DOD data communications.

Implications for Further Research

Further research must continue to refine tools to the study of large network problems. These tools must be used to investigate tradeoffs between terminal and computer density, traffic variations, the effects of improved local access schemes, the use of domestic satellites in broadcast mode for backbone networks, and the effect of link and computer hardware variations in reliability on overall network performance. The potential of these networks to the DOD establishes a high priority for these studies.

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Chapter 1
SUMMARY OF ACCOMPLISHMENTS

1.1 Introduction

Defense Department communications and computational requirements are global in scope and immensely complex. Communication problems range from the local collection and distribution of information between the components of a single unit in a small geographic area to the global transportation of strategic intelligence and command and control data. Defense Department network problems have become so complex that the unaided human cannot comprehend, let alone solve them effectively. The importance of these problems makes it imperative that effective analysis, design aids and new methods of communication be developed to meet the DOD's rapidly evolving requirements.

Network Analysis Corporation's (NAC) activities under the current contract have dealt with problems in local, regional, and large scale data communications, network analysis, design, cost, performance, and the application of computer techniques to support the analysis and design activities. NAC's efforts can be categorized into the following major areas:

- Packet Radio System Network Studies
- Packet Radio System Network Algorithms and Controls
- Local and Regional Data Network Performance and Cost Comparisons
- Integrated Large Scale Packet Switched Network Cost and Performance
- Support Facility Development

Results accomplished during the contract period in the above areas are summarized in the following paragraphs.

1.2 Packet Radio System Network Studies

Efforts during the past contract year were aimed at establishing base performance characteristics of a single station, fixed repeater location Packet Radio System and to evaluate the effects on performance of a number of fundamental hardware design decisions.

Analytic and simulation studies of throughput and delay were conducted to enable various design decisions including: use of multiple or single detectors at repeaters and stations; evaluation of tradeoffs between range, power and interference; incorporation of single or dual data rate repeaters; common versus split channel operation; and the use of directional versus non-directional antennas. In addition, numerous studies were performed to quantify system delay, throughput and blocking under various routing alternatives, acknowledgement schemes, repeater network organization and to insure that gross system performance using unoptimized operating parameters and algorithms was within a level that would justify further design efforts.

1.3 Packet Radio System Network Algorithms and Control

During the last project year, the main effort has been towards developing workable network algorithms, to insure order of magnitude performance and design robustness for a single station, multiple repeater, multiple terminal network. Preliminary designs of eleven routing algorithms were evaluated using combinatorial analysis. Three were selected for detailed design; two of these were simulated and tested, and based on these tests, recommended for implementation. Single station, multirepeater initialization, network mapping, and transmission algorithms were developed, but have not yet been simulated or tested. Hop-by-hop and end-to-end acknowledgement schemes were developed, simulated and tested. Simple terminal search and local terminal control algorithms were developed and simulated. In addition, a repeater location optimization algorithm was developed, programmed, and tested. The above family of algorithms provided a basis for demonstrating the reliable transmission of packets within the Packet Radio System, but further work is required to improve efficiency, to handle multiple stations, and to increase the number of types of terminals that can be handled by the system.

1.4 Local and Regional Data Network Performance and Cost Comparisons

During the previous contract years, a variety of tools have been developed to allow economical cost-performance tradeoff studies. Studies performed include: the practical demonstration that low cost terminal access can be achieved by hardware multiplexing at TIPs; the proof of the cost-effectiveness of multipoint lines for connecting low and medium speed terminals into ARPANET; the demonstration of the use of software demultiplexing as a means of increasing the terminal handling capacity of a TIP by a factor of 10; and the theoretical calculation of capacity, error rates and delay for a system incorporating broadcast packet radio techniques on a wideband coaxial cable local distribution network servicing a large suburban population.

1.5 Integrated Large Scale Packet-Switched Network Cost and Performance

During the contract period, the groundwork was laid to complete the study of cost and performance tradeoffs in large scale packet-switched networks. Basic analysis and design algorithms for optimization of terminal processor location, topological optimization, throughput and delay analysis, and reliability analysis have all been completed. These programs presently operate in stand alone mode and must be integrated in order to complete the large network studies. In addition, a number of cost performance studies have also been completed. These include studies of the impact of satellites on a 40 node ARPANET, the establishment of the feasibility of a 1,000 IMP packet switched network using terrestrial links, and studies of the cost-effectiveness of packet switching within an environment containing several thousand terminals. These studies are expected to lead to methods for handling large numbers of terminals and processors and various packet access methods implemented within different hierarchy levels of a large integrated C^3 network.

1.6 Support Facility Development

During the past year, a basic packet radio simulator was developed. The simulator handles a single station, up to 48 repeaters and several hundred terminals of the same type. Imbedded in the simulator are models of the repeater, station, and terminals, two routing algorithms, non-persistent carrier sense and unslotted ALOHA random access schemes, zero capture receivers, single and dual data rate channels, omnidirectional antennas, and an interactive terminal to station protocol. All device actions required to initiate, relay, and receive a packet are simulated in the same sequence of events that would occur in the actual packet radio system. Last year's experience showed that for systems like packet radio, interactive, graphical display can greatly reduce the time required to carry out certain forms of system studies such as repeater data rate, power, and operating parameter variations. During the contract period, the first phase of a graphical display system, specifically designed for handling network problems, was developed. In addition, several stand alone analysis and design algorithms and programs have been developed, including a repeater location algorithm and a basic network editor to serve as a front end to the network analysis and design programs. These have not yet been incorporated into the simulator.

The role of the packet radio simulator has been to quantify the network impact of basic device design decisions and to quantitatively demonstrate that the basic packet radio design provided technically effective and reliable packet transmission. The simulator was developed with a general data structure to allow capability for extension, but was used during the past year to deal only with a single station, multirepeater configuration. An important area of work for this year will be to extend the simulator to incorporate up to ten stations.

1.7 Emphasis of This Report

In this report, we summarize the principal results obtained during the contract period. Primary emphasis in this document is placed on the system issues and tradeoffs identified and examined, rather than on the specific algorithms and models developed to evaluate them. Algorithms, models, and computer aided design tools have been extensively described in NAC's Semiannual Reports 1-3 to ARPA.

Chapter 2
MODES FOR DIGITAL NETWORKING

The late 60's and the 70's have seen the proposal and development of an incredible array of digital services. The field is, of course, still dominated by the giants of the common carriers, AT&T and Western Union.

2.1 Common Carriers

AT&T offers a wide array of digital services, many with a long history of development and usage—others in the proposal stage. A short list is given to indicate the types of bandwidths and tariff structures available [34].

- Series 8000 offers up to 48 Kbps for data transmission. The total cost is determined by mileage and service terminal costs.

- Series 5000 (Telpak) type 5700 accommodates a 240 Kbps data rate, type 5800 accommodates a 1000 Kbps data rate. For data transmission, the total bandwidth can be divided into subchannels of the desired bandwidths. The charge consists of a mileage charge and a service terminal charge.

- High-Low (HiLo) density tariff was recently approved by the FCC. Approximately 370 locations are defined to be high density traffic points; the remaining are low density traffic points. For a half-duplex channel, the following interexchange mileage rates apply:

High point-high point:	.85 $/Mile	Mo.
High point-low point or low point-low point:	2.50 $/Mile	Mo.
Short haul (25 miles):	3.00 $/Mile	Mo.

Monthly channel terminal charges are $35 for High and $15 for Low; station terminal charges are $25 for both High and Low.

A low to low connection can be implemented either directly (in which case the low to low direct distance tariff applies) or through two intermediate high density points (in which case different tariffs apply to different segments).

- Digital Data Services (DDS). This is a proposed new data service based on the T1 digital carrier. DDS will initially interconnect 24 major cities and will be progressively extended to the 370 high density traffic cities. FCC approval has been granted for Boston, New York, Philadelphia, Washington, D.C., and Chicago. Cost is based on mileage and service terminal charges which depend on the bandwidth of 2.4, 4.8, 9.6, or 56 Kbps. The T1 carrier will handle up to 1.544 Mbps. The T2 system will operate at four times that rate and the T5 now in the experimental stage, will handle 564 Mpbs on coaxial cable or waveguide.

- Picturephone, although not intended for digital services, does have a bandwidth of 6.312 Mbps and could be a vehicle for the transmission and visual display of data.

Western Union [6] is introducing a hierarchy of time division multiplexing which can either operate over digital radio paths or use modems over analog radio systems. The 1.544 Mbps and 6.3 Mbps channels are designed to be compatible with Bell System T1 and T2 lines respectively. Similarly, the 2.4, 4.8, 9.6, and 56 Kbps speeds are designed to be compatible with anticipated telephone company offerings. Electronic data switches have and are being installed to form a common switching plant for Telex, TWX, and DATA Switching.

2.2 Specialized Common Carriers

A development of crucial interest to the computer industry is the growth of a new class of common carriers--the Specialized Common Carrier. Only a few are listed here. More are appraised by Gaines [25].

Microwave Communications, Inc. (MCI) was a pioneer among the specialized carriers, filing its first common carrier application in 1963. Channel bandwidths of 1,000 Kbps are available and will include 81 cities on both coasts as well as the initial configuration of Chicago, St. Louis, Cleveland, Detroit, Toledo, South Bend, and Pittsburgh. MCI will offer multi-address distribution capability and store-and-forward capability. The total line charge is the sum of the intercity mileage charge, a system access charge, and a channel termination charge, all depending upon bandwidth required.

Data Transmission Company (DATRAN) has innovated in the concept of nationwide digital transmission since 1968 [59]. In April, 1973, the company began construction of its network. The network will offer digital communications between 35 major cities. Users within 50 miles of each such city can be connected to the network by DATRAN facilities. Services will be provided from 2.4 Kbps through 1.344 Mbps. Circuit switching will be used with connection times of less than .5 seconds. Network rates will be similar to DDS in structure.

A number of regional common carriers originally constructed for video transmission are entering the picture as possible components of nationwide digital networks. As one example among several, Western Tele-Communications, Inc. (WTCI) began as a video microwave carrier carrying signals for CATV systems owned by its parent company, Tele-Communications, Inc. in the Western half of the U.S. It is one of the largest non-telephone microwave common carriers in the country, with over 13,000 route miles of microwave links in service. WTCI plans to convert some of its bandwidth to digital links with data circuits up to 1.544 Mbps operated on a store-and-forward basis.

2.3 Value Added Networks (VAN's)

Value added networks are communication service companies which lease transmission facilities from common carriers or specialized carriers and resell communication services not available from the original carrier. One of the services to be offered is packet communication with service comparable to that in the ARPANET. FCC approval has been granted to Graphnet, Packet Communications, Inc. and Telenet. The introduction of VAN's services may be quite rapid since they do not undertake the construction of transmission lines. Tariffs have not been determined yet, but at least for packet switching networks, the rates will be independent of distance and proportional to traffic volume.

2.4 Satellite Communications

A number of companies have applied for FCC approval to sell commercial satellite services including Amersat, CML, and WU. Some are launching their own satellites. On April 13, 1974, Western Union's Westar 1 became the first U.S. domestic communications satellite launched into orbit. Competing will be systems operated by GTE, RCA, and ATT. Their ground stations and planned operations are part of the layman's day to day information as in the map of planned ground stations from the April 15, 1974, edition of the *New York Times* (see Figure 2.1).

Others are leasing channels on existing satellites. For example, American Satellite Corporation, formed in 1972 as a joint venture of Fairchild Industries and Western Union International has leased three transponder channels on Telesat-Canada's ANIK-2 satellite with plans for earth station locations in New York, Los Angeles, Chicago, and Dallas at data rates up to 60 Mbps.

Two general characteristics of satellite rates are insensitivity to distance with a strong volume discount for use of satellite bandwidth.

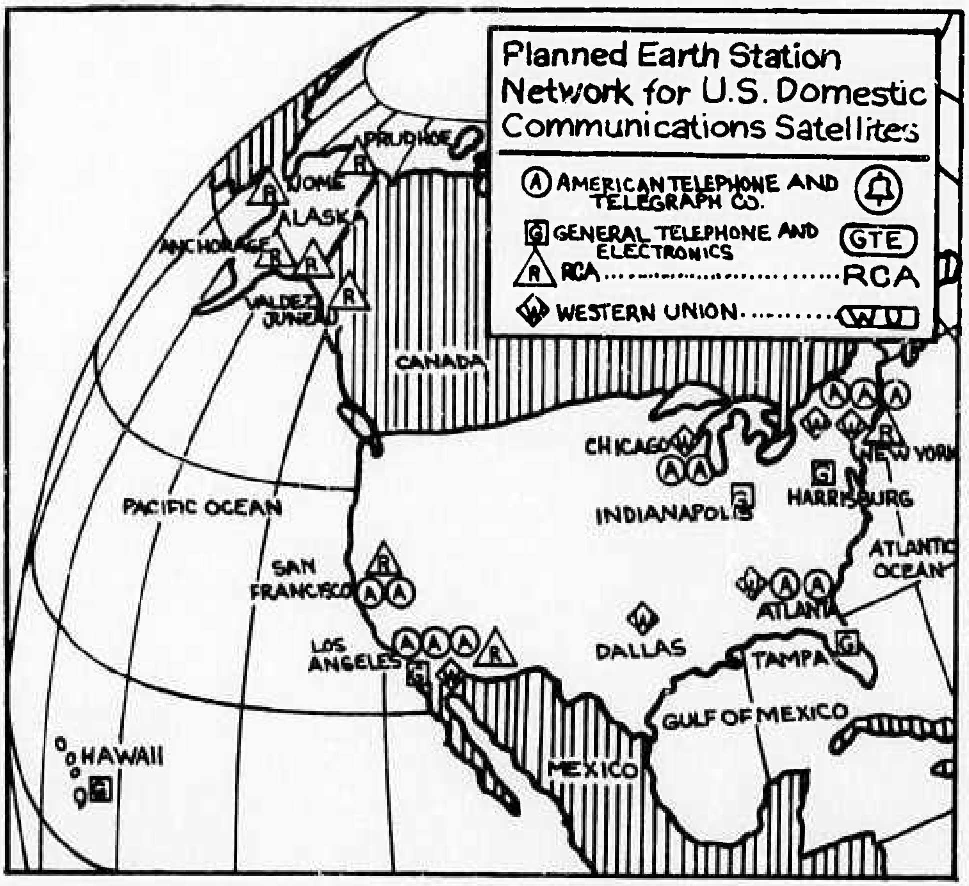


Figure 2.1: Satellite Map of U.S.

Chapter 3
NETWORK CONFIGURATION

The abundance and variety of services provided by the common carriers and specialized common carriers could mean low cost data transmission involving novel interconnection of facilities to match a broad range of user requirements, or it could mean failing networks and poorly spent dollars.

Let us first say that the main danger is not among the more classical problems of interconnecting facilities. There are, of course, challenging problems in interfacing terminal equipment from a range of manufacturers with a cascade of networking services. Hardware and software design for interfaces is required to assure proper synchronization, compatible message structure, electrical matching, and a host of other functions. These are important but, within the context of offered services, predictably solvable problems. The problem of servicing and maintaining a multilinked network with multiple owners is often a frustrating one with each vendor blaming a system outage on the others. But, this is also a solvable management problem.

The potential for difficulties is the complexity of configuring the best network to meet requirements and in designing the best method to locally access the network. If these tasks are improperly handled, the resulting networks may be more expensive than conventional approaches. Also, they might fail! Time delays in an interactive system might be so long that a programmer at a keyboard loses patience and abandons the system; times for bulk file transfers may become so long that the user opts for air express instead; users may receive so many busy signals that they lose business.

In both the national network and the local network, the user must consider the full gamut of constraints including throughput, delay [17, 12], reliability [14, 54, 37], and cost, among many others. These may affect not only network configuration, but also selection of mode of transmission and of the carrier.

For example, one of the properties of the ARPANET, and therefore, presumably of VAN's networks, is a high degree of reliability because of alternate routing and retransmission of lost packets. The varying requirements and performance profiles of different transmission schemes will dictate the use of sophisticated analysis and design aides. The characteristics of these aids are described below. The primary technical problem is the selection and configuration of the facilities to meet requirements, and to achieve savings. It is, at present, not feasible to develop a computer program which will optimize based on all commercial offerings. Nevertheless, it is clear to anyone who has tried to optimize even a simple problem involving only Telpak rate structures that network design must be computerized. Of necessity, the design programs must be characterized by an extreme degree of flexibility.

a. Programs must be modularized. Subnetwork designs, characterized by geographic locations, functions or traffic must correspond to modules of the program so that changes in tariffs and grade of service require only localized program modifications. Furthermore, new design methods can be achieved with rearrangement of modules.

b. Network design programs must be parametric. The designer must be able to evaluate requirements based on performance and cost. Traffic, delay, and throughput requirements can never be specified precisely *a priori,* but must be selected on the basis of tradeoffs of cost versus performance.

c. Programs must be capable of performance evaluation. From traffic statistics and measurements, the manager must be able to determine when a network must be upgraded.

d. Design programs must be interactive. It has been a common experience in many areas of operations research, combinatorics, and large scale network analysis and design that the human is an important element in the design loop.

e. Computational efficiency must be given paramount consideration. An improperly structured data base, a failure to consider sparsity, or an inefficient combinatorial subroutine can cause serious degradation in network performance and cost design.

Let us consider item a, modularized programs, as an example. In *Semiannual Report No. 2,* we gave a classification of general cost structures which includes most commercial tariffs and enables the development of efficient modules for each structure.

- Distance Dependent (DID) structures. The cost per channel between two points is a function only of the distance between the points; it is independent of the specific locations of the points or the bandwidth. Examples of DID structures are the AT&T type 8000 tariff and the Hi Lo density tariff, where both points are high density points.

- Location Dependent (LOD) structures. The cost per channel between two points depends on the location of the points, but not on the bandwidth. A typical example is the Hi-Lo density tariff if at least one point is a low density point. Another example is VAN's for non-network points.

- Volume Discount (VOD) structures. The cost of leasing additional bandwidth between two points decreases with the number of channels already leased. The cost also depends on the distance between the points, but not on their locations. Examples are the Telpak tariffs (see Figure 3.1), the DDS tariff for network points, the specialized carriers, satellite companies, and VAN networks. The case in which the line cost depends on terminal locations as well can often be reduced to the study of two separate LOD and VOD problems.

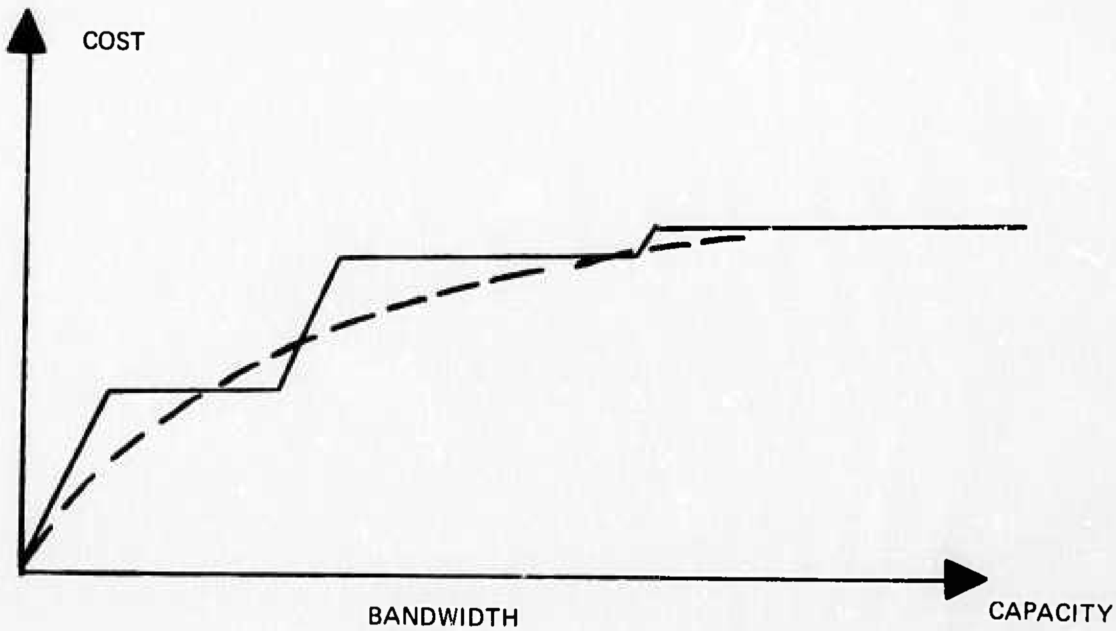


Figure 3.1: Telpak-Like Tariff

Each of the tariff structures is amenable to a different set of analytical and heuristic methods. For example, under VOD structures, network design tends to route traffic on links that provide the best economies of scale. Generally, cost structures with strong economies of scale lead to sparsely connected or tree topologies while structures with weak economies of scale lead to highly connected topologies. The network algorithms for VOD problems generally compute shortest routes according to appropriate link costs which vary with each iteration [59, 19, 13]. The specific algorithms and their efficiency depend upon the specific cost capacity function of the links [15]. Link functions which are neither convex nor concave and have large capacity jumps such as the Telpak-like tariff below, require very complex combinations of analytic and heuristic methods [47]. On the other hand, continuous concave cost-capacity functions can be handled exactly using mathematical programming techniques [13].

To illustrate how these modules are combined to solve a problem, we consider the design of a national network consisting of a terrestrial VAN and satellite links with private ground stations. Given the traffic requirements and tariffs, we wish to accommodate the traffic requirements with a specified grade of service at minimum cost. The network shows a strong economy of scale due to volume discount for satellite bandwidth and the high cost of ground stations. But, there is a complex tradeoff between terrestrial line cost, satellite bandwidth cost, and ground station cost.

A possible procedure for this problem begins with picking the number and location of ground stations (these may be varied for different designs). To connect non-network points to the VAN's terrestrial network we assign an LOD cost structure. We then satisfy node pair requirements along minimum cost routes relative to the LOD structure. From this assignment we determine national network requirements. Using a VOD structure, we then design the national network using the link costs for terrestrial and satellite links. Suboptimization of the terrestrial network is possible here. Once the nationwide network is obtained, the LOD cost structure is recomputed based on marginal costs and the procedure is iterated.

To solve such networks problems, which are quite large, will require special techniques that have not yet been developed. In the following chapters, we describe various network problems, ranging from cost and performance of large distributed networks to the control of traffic in a local broadcast packet network. The goal of the discussion will be to indicate the status of each research area and the open issues.

Chapter 4
LARGE SCALE DISTRIBUTED NETWORKS

4.1 Introduction

Performance of a distributed computer communication network is usually characterized by the parameters of cost, throughput, response time, and reliability. Analysis and design of large scale networks requires techniques substantially different from the ones used for smaller networks. Similarly, implementation in the actual network, of procedures such as routing and flow control are significantly impacted by network growth, and implementations suitable for a 50 node network may be totally inappropriate for a 500 node network required to perform the same functions.

Network design must be concerned with the properties of both the node's and the network's topological structure.

The outstanding design problem for large distributed networks is to specify their routing and topological structures. This specification must make full use of a wide variety of circuit options. Preliminary studies indicate that, initially, the most fruitful approaches will be based on the partitioning of the network into regions or, equivalently, constructing a large network by connecting a number of regional networks. In addition to reducing the computational complexity of the topological design problem, nodes may be clustered into regions for numerous reasons, such as:

- To partition status information for use in routing, flow control, and other decision processes within the operating network

- To determine regions of low-, medium-, and high-speed lines in hierarchical structures

- To find concentrator-multiplexer locations

To send a message in a distributed network constructed by connecting together a set of regional networks, a node might specify both the destination region and the destination node in that region. No detailed implementation of a large network has yet been specified, but an early study of their properties indicated that factors such as cost, throughput, delay, and reliability are similar to those of the present ARPA Network, if the ARPA technology is used [10].

One of the first questions that arises in the decomposition problem is how large to make the regions and how many regions to create. In a multilooped network, each loop would define a region. Once the number of regions has been determined, nodes must be assigned to each and interface nodes selected. This problem is extremely difficult, and little is presently known about this "clustering" problem.

Before any substantial progress can be made in applying existing clustering techniques to large computer networks, an appropriate "distance" or "nearness" measure must be selected, probably on the basis of intuition and experiment.

For the design of large computer networks, clustering requires the assignment of distance measures to take into account cost, capacity, traffic, delay, reliability, and routing. Almost no general theoretical results are presently known for this problem.

In this section, we report the results of a sequence of experimental network designs aimed at investigating the economic and performance tradeoffs of distributed computer communication networks as a function of size. Three sequences of designs were performed: 20-100 nodes, 200 nodes and 1000 nodes. The first series of networks were thoroughly optimized using the optimization techniques discussed [11]. The 200 node networks were partially optimized (within the limitations of a small finite computer time budget), while the 1000 node network designs represent workable network designs whose structure was chosen for both buildability and mathematical tractability.

4.2 The Network Model

The network model chosen for the study was the ARPANET [18]. This system, which employs packet switching is likely to be the prototype for most future distributed computer communication networks.

4.2.1 Message Handling

The message handling tasks at each node in the network are performed by Interface Message Processor (IMP) located at each computer center. The centers are interconnected through the IMP's by fully duplex communication lines. When a message is ready for transmission, it is broken up into a set of packets, each with appropriate header information. Each packet independently makes its way through the network to its destination. When a packet is transmitted between any pair of nodes, the transmission IMP must receive a positive acknowledgement from the receiving IMP within a given interval of time. If this acknowledgement is not received, the packet will be retransmitted, either over the same or a different channel depending on the network routing doctrine being employed.

4.2.2 Design Goals

A design goal of the system is to achieve a response time of less than 0.5 seconds for short messages. The final network must also be reliable, and it must be able to accommodate variations in traffic flow without significant degradation in performance. In order to achieve a reasonable level of reliability, the network must be designed so that *at least* two nodes or links must fail before it becomes disconnected. For small networks, this design goal ensures adequate reliability but for larger networks, additional provisions must be made [55]. Thus, the 1000 node design was specially handled with respect to its reliability considerations.

4.2.3 Routing Procedure

For the network designs containing 200 or fewer nodes, the *Minimum Node Routing* procedure described in [13] was employed. However, for the 1000 node network design, the *Cut Saturation Technique* described in [4] was used. This improved procedure was used in an attempt to compensate for the lack of optimization of the 1000 node network topology.

4.2.4 Cost Structure

Cost structures for lines and IMP's are shown in Tables 4.1 and 4.2. Note that the line costs are those available to the U.S. Government under the Telpak tariff.

Table 4.1: (all lines FDX)

Capacity (Kbps)	Data Set Cost/Month	Line Cost Per Mile/Month
9.6	$ 493	$ 0.42
19.2	$ 850	$ 2.50
50.0	$ 850	$ 5.00
230.4	$1300	$30.00
1544.0	$2000	$75.00

Table 4.2: Message Processor Cost

Description	Purchase Cost	Cost/Year*
DDP-316 IMP (Max throughput = 600 Kbps)	$ 50,000	$15,000
DDP-516 IMP (Max throughput = 800 Kbps)	$ 70,000	$21,000
DDP-316 TIP (Max throughput < 600 Kbps)	$100,000	$30,000
HSMIMP (Max throughput = 6,000 Kbps)	$250,000	$75,000

*Yearly cost is assumed 30% of purchase cost

4.2.5 Node Locations

Node Locations strongly influence network efficiency. For example, at one traffic level a 20-node network may be more efficient than the same network with two additional nodes while it may be less efficient at other traffic levels. Consequently, a systematic and hopefully "unbiased" procedure is required to generate realistic systems with differing numbers of nodes.

Many location algorithms are possible. For example, nodes could be located on the basis of any of the following requirements: industrial concentration; distribution of military bases; distribution of universities; population.

Since both industry and universities tend to be located at population centers, distributions produced by these factors are positively correlated. On the other hand, military bases are often located at a distance from population centers, and hence such nodal distributions would have a negative correlation with the others.

For the present study, nodes were located on the basis of population. Within each metropolitan area, IMP's were assigned, 20 miles apart, in proportion to the population of the area. As examples, the 20 node design connected 10 metropolitan areas, the 100 node design connected 40 metropolitan areas, the 200 node design connected 62 metropolitan areas, and the 1000 node design connected 238 metropolitan areas.

4.2.6 Traffic Assignment

A fundamental problem in all network design is the estimation of the traffic the network must accommodate. For some problems, accurate estimates of user requirements are known. However, complete studies are not yet available to predict the flow requirements in networks of the type being considered here. A number of basic questions are yet to be resolved. For example, it may be reasonable to assume that the flow out of a node will be proportional to the population assigned to that node. However, will the flow between two nodes be affected by the distance between these nodes? If so, how will the cost-throughput characteristics of the network be affected?

In order to investigate the effect of different traffic distributions on network economy, a sequence of experiments described in [10], was conducted in which traffic patterns were varied as a function of distance and low cost networks for these patterns generated. These experiments indicated that networks with comparable costs could be designed for widely varying traffic requirements. Hence, in the design experiments used to generate cost-performance tradeoffs, equal traffic requirements between all nodes was assumed.

4.2.7 Summary of System Parameters and Characteristics

The following is a summary of the factors utilized in the network design.

a. The system contains message processors located in the largest cities of the Continental United States. The number of message processors in each metropolitan area is proportional to the population of the town.

b. Required traffic between message processors is assumed uniform for all node pairs. Traffic levels in the range from 3 to 20 Kbps/node are considered.

c. Messages are assumed to have the same structure and formats as in the present ARPANET configuration. Message delay is evaluated for single packet messages.

d. The nominal traffic level is set at 80% of the saturation level in order to maintain within acceptable limits the queue size of packets awaiting transmission on each channel.

e. The link failure rate is assumed equal to 0.02. The node failure rate is assumed equal to 0.02 for IMP and TIP processors, and .0004 for redundant configurations (IMP or TIP plus backup, or redundant high speed modular IMP configurations).

f. The high throughput presented by a 1000 node network requires very high channel and message processor rates. Therefore, in the design, two high speed hardware options—the 1544 Kbps data channel and the HSMIMP (High Speed Modular IMP)—have been considered in addition to the options already available. Such high rate options are presently under development and will soon be operational offerings.

4.3 Regional Partitions and Hierarchical Network Structure

The network design is based on a regional decomposition principle. The system is divided into regions. Nodes are then uniquely assigned to each region. These nodes are classified as either exchange nodes or local nodes. There may be any number of regions, and the choice of local and exchange nodes is made by the designer. The distinction between nodes stems from restrictions on allowable connections.

a. Within a given region any connection is possible.

b. Connections between regions are only allowed between exchange nodes.

c. Any connection between exchange nodes is possible.

For the designs containing no more than 200 nodes, a two-level regionalization is used. For these networks, the network connecting the exchange nodes is called the national network while the networks connecting local nodes are called regional networks. Therefore, a flow from a local node in region I to a local node in region J must be routed first through the regional network to an exchange node in region I, then through the national network composed of exchange nodes to an exchange node in region J and finally through another regional network to the destination node in region J. Thus, in general, a hierarchical network is designed. If desired, structural constraints can be eliminated by declaring every node to be an exchange node. In this case, there are N(N-1)/2 possible links where N is the number of nodes in the network. If N is large, the computational time required for optimization may be prohibitive, and therefore, decomposition is essential. For the sizes of the networks being considered, the decomposition approach produces computation time savings ranging from a factor of ten to factors of more than 400.

The determination of the optimal topology in a 1000 node hierarchical network is a very complex problem since it requires the solution of a large number of subproblems, all related to one another. For example, one must optimally determine:

- The number of hierarchical levels
- The node partitions
- The topology within each partition
- The connections between networks in different levels of the hierarchy

Because of the complexity of design optimization, only feasible, reasonably low cost designs were considered in the 1000 node study. A feasible design in fact is sufficient for the determination of cost, throughput, delay and reliability trends with respect to network size, and for a comparison between hierarchical and non-hierarchical structures.

The hierarchical structure (see Figures 4.1, 4.2 and 4.3) here considered consists of *three* hierarchy levels: one 10 node national network, ten 10 node regional networks, and one hundred 10 node local networks. Each local network is considered as one "node" of the higher level regional net, and similarly, each regional net is one node of the national net. Various ways of connecting lower to higher level networks can be considered. In the cost/throughput study, we assume for simplicity that each subnetwork communicates with the higher level network only through one "exchange" node. In the reliability study, however, two and three exchange node configurations are also considered.

4.4 Reliability

Reliability constraints play increasingly more important roles with growth of the network [55]. For example, in the early stages of the ARPANET growth, adequate reliability was

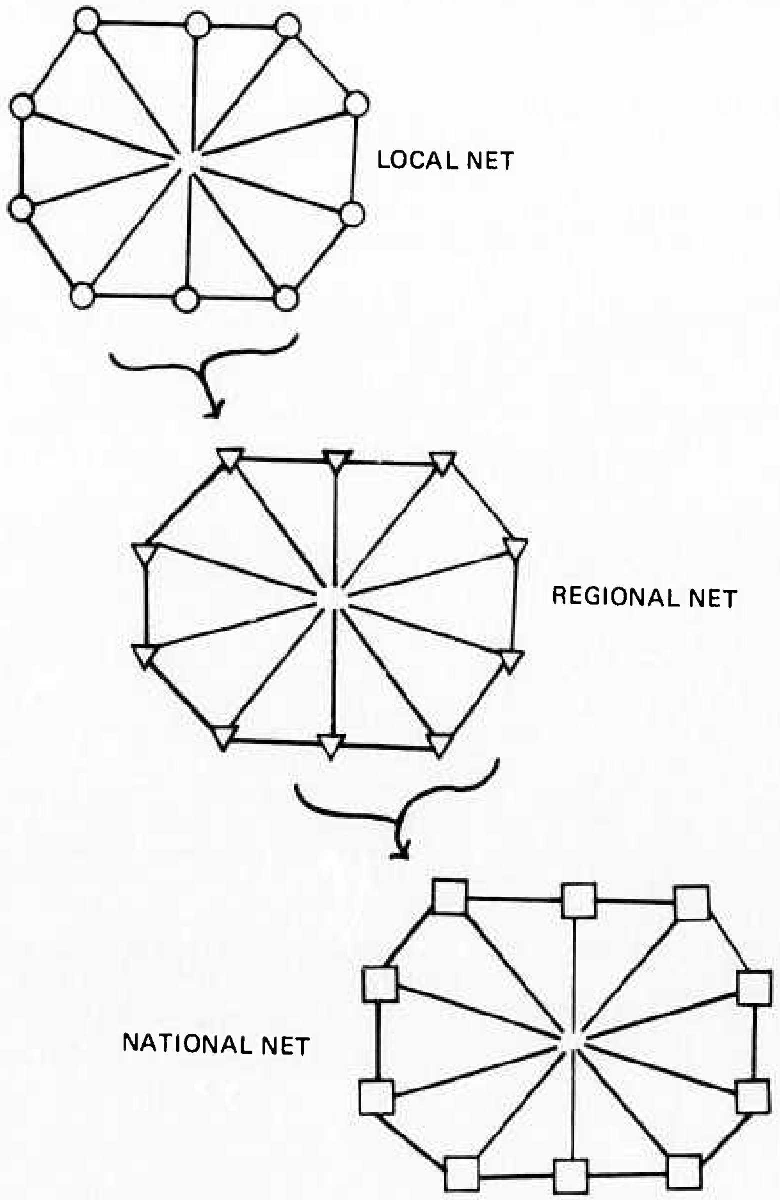


Figure 4.1: A 1000 Node Network Composed of 10 Ten-Node Regional Nets Each Containing 10 Ten-Node Local Nets

achieved by the provision of two node disjoint paths between all pairs of nodes. However, even at its present size (approximately 40 nodes) this is no longer a sufficient guarantee of adequate network reliability.

Network reliability analysis is concerned with the dependence of the reliability of the network on the reliability of its nodes and links. Element reliability is easily defined as, for example, the fraction of time the element is operable, or as by the mean time between failures and expected repair time. The proper measure of network reliability is not as

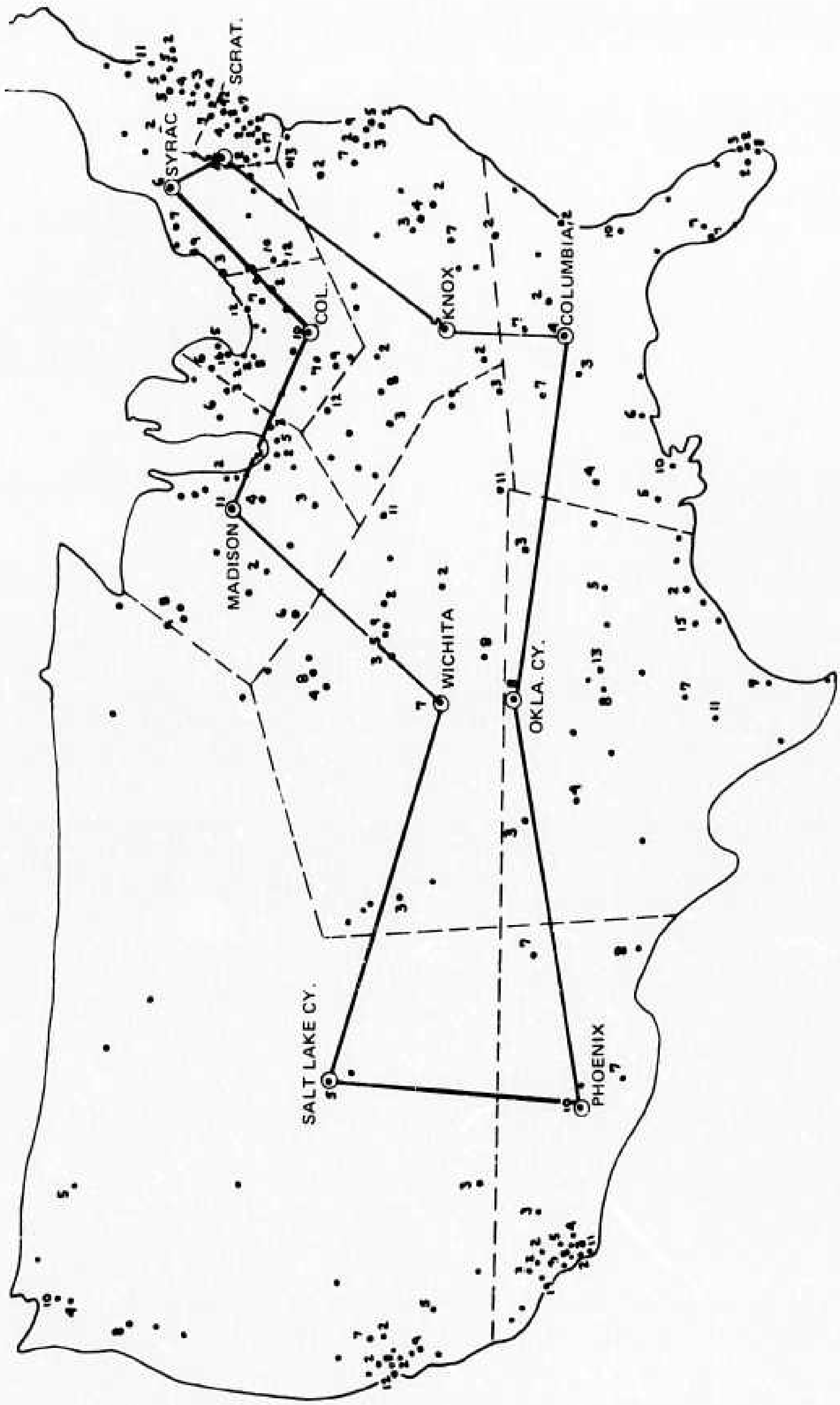


Figure 4.2: National Network and Regional Partitions

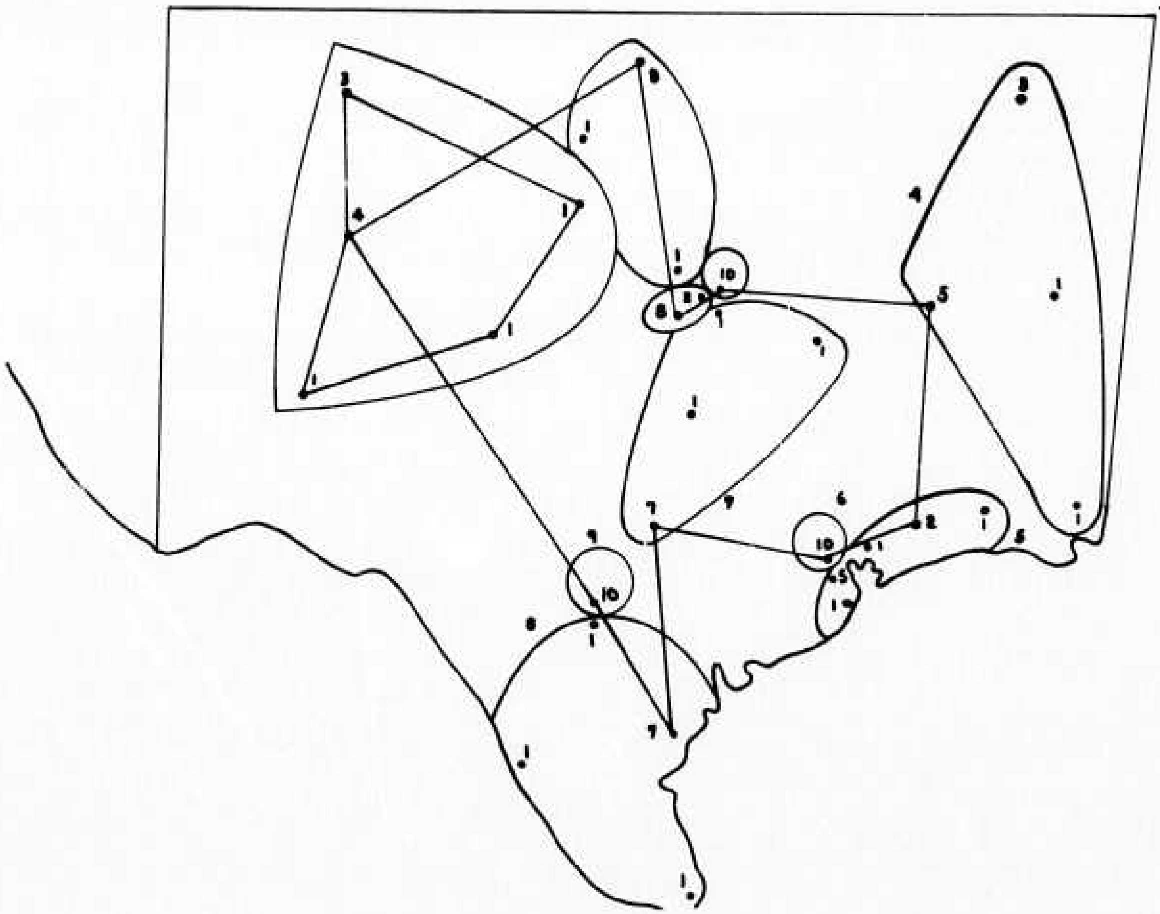


Figure 4.3: Regional Network; Local Partitions; Local Network

clear and simple. Three possible measures are: the number of elements which must be removed to disconnect the network, the probability that the network will be disconnected and the expected fraction of node pairs which can communicate through the network. Many other measures can and have been suggested.

Figure 4.4 shows the results of reliability analyses of a set of 20-100, and 200 node networks designed to meet throughput requirements of approximately 8 Kbps/node under the assumptions that nodes are perfectly reliable. These networks were designed with a "two connectivity" constraint and were optimized to provide the required throughput at least cost. As is evident, the larger networks are significantly less reliable than the smaller networks. Thus, extension of the same design principle to the 1000 node design would be likely to lead to a low reliability system. Therefore, the 1000 node network is considered under varying conditions of backup and structure. These changes are made at the *local* level. The regional and national levels are always three connected.

To evaluate the reliability of the hierarchical 1000 node network, we make the assumption that two nodes in the same subnetwork can communicate with each other only through

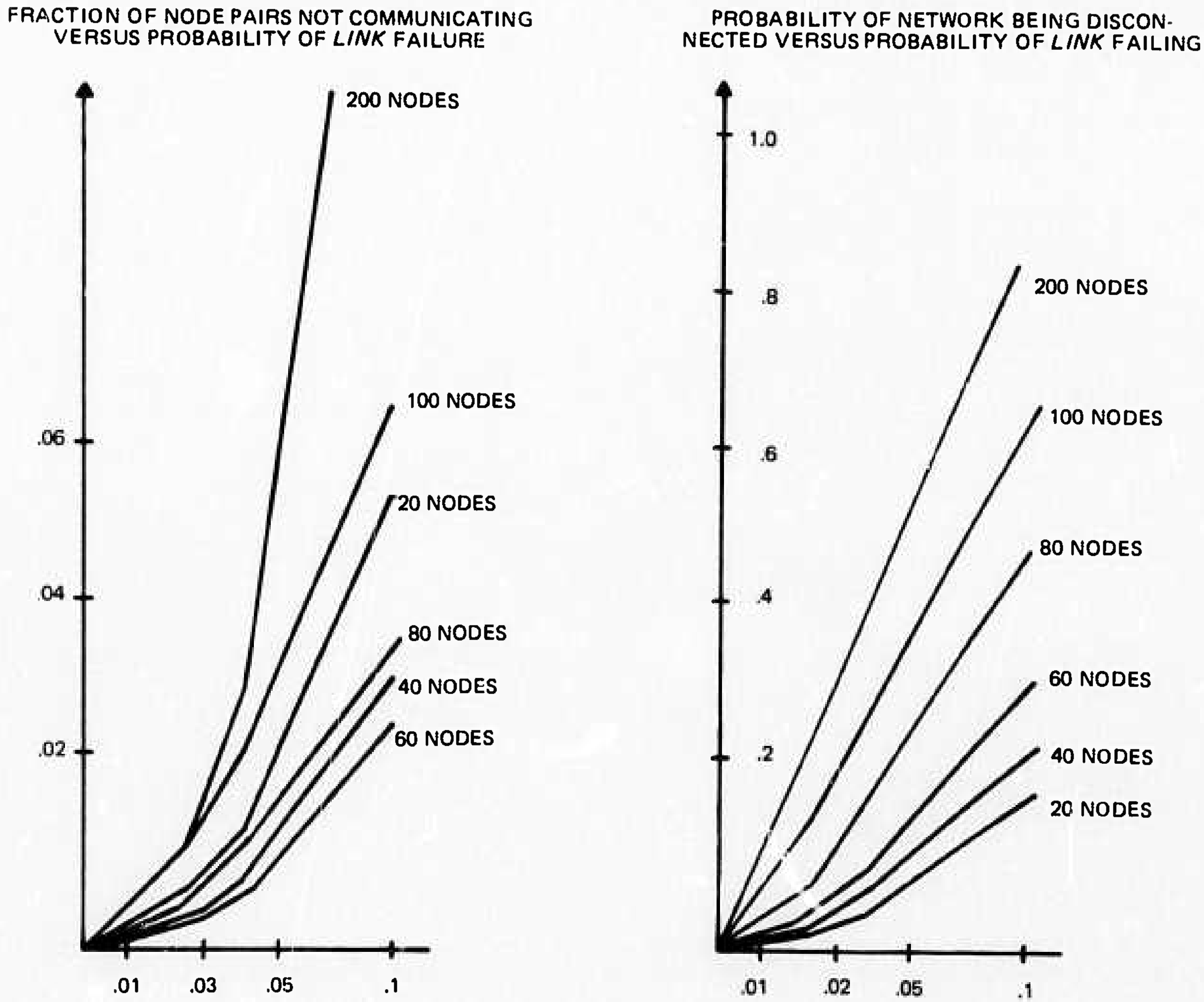


Figure 4.4: Network Reliability As A Function of Number of Nodes

paths entirely contained in the subnetwork. Therefore, two nodes of the same subnetwork can be disconnected even if there is a connection path through the higher level network. This assumption is very realistic because, in a hierarchical routing implementation, the capability of sending local or regional traffic along paths external to the corresponding local or regional net, can be achieved only with considerable increase in complexity and overhead of the routing algorithm.

With the above assumption, the probability P_{nt} of the total network being disconnected is given by:

$$1 - P_{nt} = (1-P_{n\ell})^{100} \times (1-P_{nr})^{10}$$
$$\times (1-P_{nn}) \times (1-P_{ex})^{110}$$

where $P_{n\ell}$ = probability of local net disconnected

P_{nr} = probability of regional net disconnected

P_{nn} = probability of national net disconnected

P_{ex} = probability of exchange node (or nodes) failure, which isolates the corresponding subnetwork

To evaluate F_{nt}, the fraction of disconnected node pairs, we make the simplifying (and conservative) assumption that whenever a subnetwork becomes disconnected, only one half of the nodes in the subnetwork can communicate, on the average, with the exchange node (or nodes). With such an assumption, if we let N be the number of nodes in the local net (in our case N = 10) and a_ℓ, a_r, a_n the number of noncommunicating node pairs resulting from the disconnection of a local, regional or national network respectively, we have:

$$a_\ell = \frac{N}{2}\left(N^3 - \frac{N}{2}\right) P_{n\ell} + N(N^3 - N)\, P_{ex}$$

$$a_r = \frac{N^2}{2}\left(N^3 - \frac{N^2}{2}\right) P_{nr} + N^2 (N^3 - N^2)\, P_{ex}$$

$$a_n = \frac{N^3}{4} P_{nn}$$

If we make the assumption that the above contributions are statistically independent of one another, then we can sum them up and obtain the following expression for F_{nt}:

$$F_{nt} = 2P_n(1-P_n) + \frac{2}{N^6}\left\{ a_\ell N^2 + a_r N + a_n \right\}$$

$$\cong 2P_n(1-P_n) + P_{n\ell} + P_{nr} + \frac{P_{nn}}{2} + 4P_{ex}$$

where P_n is the node failure rate, and $2P_n(1-P_n)$ is the fraction of disconnected node pairs resulting from source and/or destination failures.

To evaluate P_{nt} and F_{nt} as from expressions (1) and (5), we need to know the network disconnection probability P_{nc} for the basic, 3-connected 10 node structure. The following results were obtained using the reliability analysis programs described in [17, 12]:

$$P_{link} = .02;\ P_{node} = .02 \rightarrow P_{nc} = 7.10^{-4}$$

$$P_{link} = .02;\ P_{node} << .02 \rightarrow P_{nc} = 8.10^{-5}$$

To test the effect of various topological and back-up conditions, P_{nt} and F_{nt} are evaluated for a variety of network configurations which differ in:

1. Number of exchange nodes

2. Redundancy in the exchange nodes

3. Connectivity of the local network

Figure 4.5 illustrates the various configurations and Table 4.3 summarizes the results.

4.5 Cost-Throughput Trends

An important point which must be emphasized about the results to follow is that these results present a conservative picture of the relationship between cost and throughput. There are two major reasons for this.

Each point represents a *feasible* network obtained by either the computer network design program or by specification of the 1000 node network topology. Thus, to generate the specified throughput, no greater cost would be involved. However, because of the number of points needed to generate adequate curves, it is prohibitively costly to devote a large amount of computer time to optimize completely each design point. Therefore, if a specific throughput were to be required, a more intensive optimization would be warranted and a lower cost design would be probable.

In each design, only hardware and line options available now or in the near future have been allowed. Other developments could substantially reduce costs. For example, in [18] we demonstrated the economics created by using a 108 Kbps data set. Although this data set was once tested by AT&T, it is not a commercial offering. However, the costs involved in building a large computer network could justify the independent development of such a data set.

To illustrate the tradeoffs that occur, we first examine the 1000 node network.

For the 1000 node topology, total cost and delay for a given throughput can be obtained by analyzing 111 subnets and properly combining the results. Such an extensive analysis is too cumbersome in our case since we are interested in using throughput as a parameter. Therefore, to simplify the computation, only the costs of the national network shown in Figure 4.2 and the regional and local nets shown in Figure 4.3 were thoroughly computed, and the results interpreted as representative for all other regional and local nets. Notice that the above approach generates imprecision in the total cost, but provides the correct answers for both delay and throughput.

Figure 4.6 shows cost, throughput and delay of the *national* network for three different capacity allocations. The lowest cost configuration uses all 230.4 Kbps channel capacities. The intermediate configuration uses 1.544 Mbps channels for the outer loop, and 230.4 Kbps for the cross links. The highest cost configuration uses all 1.544 Mbps channels.

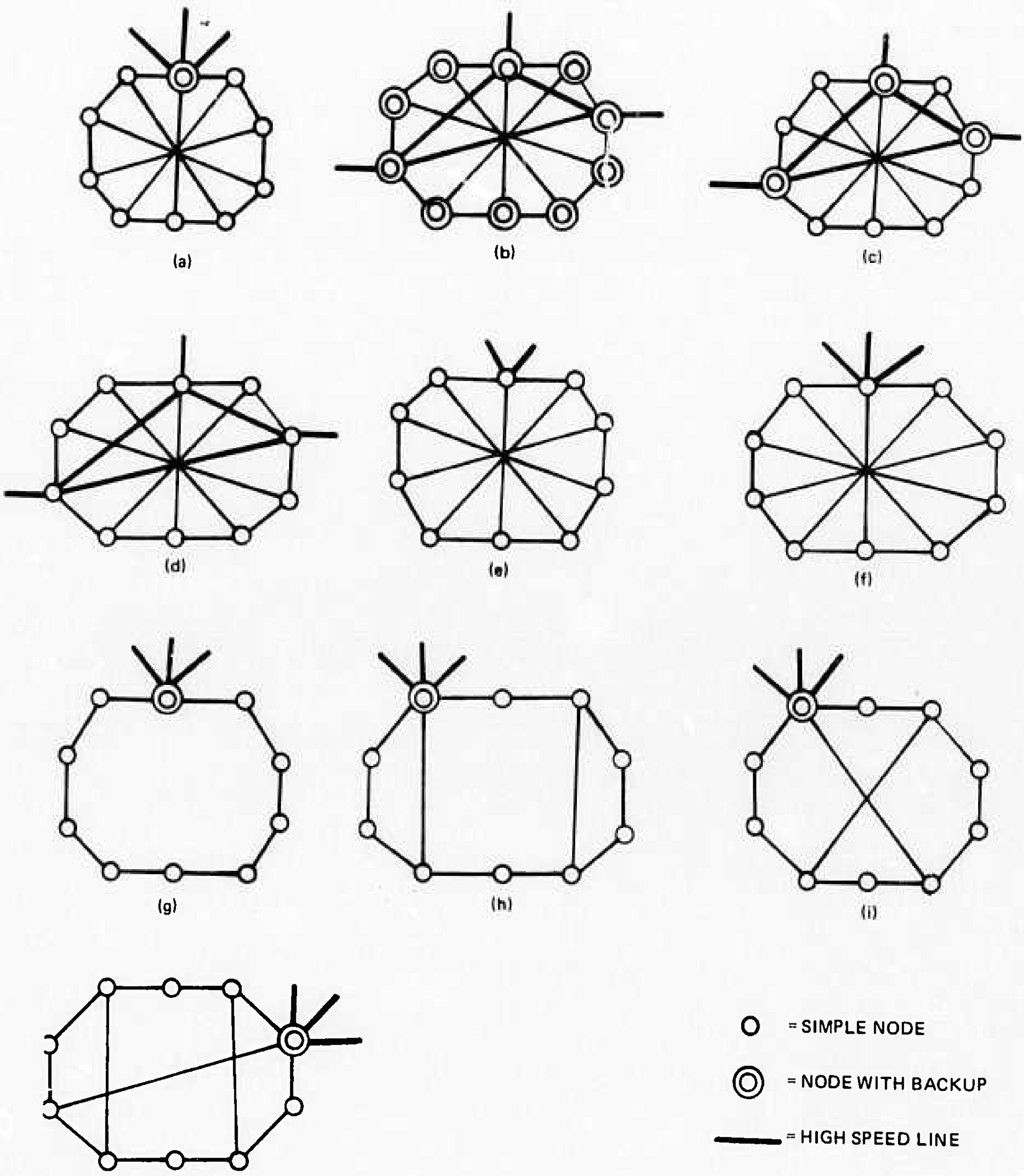


Figure 4.5: Various Local Network Configurations

Table 4.3: Failure Probabilities for Differing Networks of Figure 4.5

Network	Number of Exchange Nodes	Location of Backup	Disconnection Probability	Average Fraction of Disconnected Node Pairs
a	1	all exchanges	.011	.042
b	3	all nodes	.009	.001
c	3	all exchanges	.078	.028
d	3	none	.095	.044
e	2	none	.128	.044
f	1	none	.9	.12
g	1	all exchanges	.999	.114
h	1	all exchanges	.93	.066
i	1	all exchanges	.84	.058
j	1	all exchanges	.65	.05

The throughput, expressed in Kbps/node, refers to the local nodes; therefore, the throughput of each of the 10 "supernodes" in the national net is approximately 100 times higher. The cost in Figure 4.6 reflects line and data set costs. The additional message processor cost is now evaluated, assuming that each node has redundant processors:

a. Lower cost net:
20 x DDP-316 IMPs, cost = .3M$/Year

b. Intermediate cost net:
20 x HSMIMPs, cost = 1.5M$/year

c. Higher cost net:
20 x HSMIMPs, cost = 1.5M$/year

Figure 4.7 shows the results for the *regional* net. The lowest cost solution uses mostly 50 Kbps channels; the highest cost solution includes several 230.4 Kbps and 1.544 Mbps channels. The throughput refers to local nodes. Assuming that each node has redundant processors, the message processor cost is given below:

a. Lower cost net:
18 x DDP-316 IMPs, cost = 270K$/year

b. Intermediate cost net:
16 x DDP-316 IMPs, cost = 240K$/year
2 x HSMIMPs, cost = 150K$/year
Total cost = 390K$/year

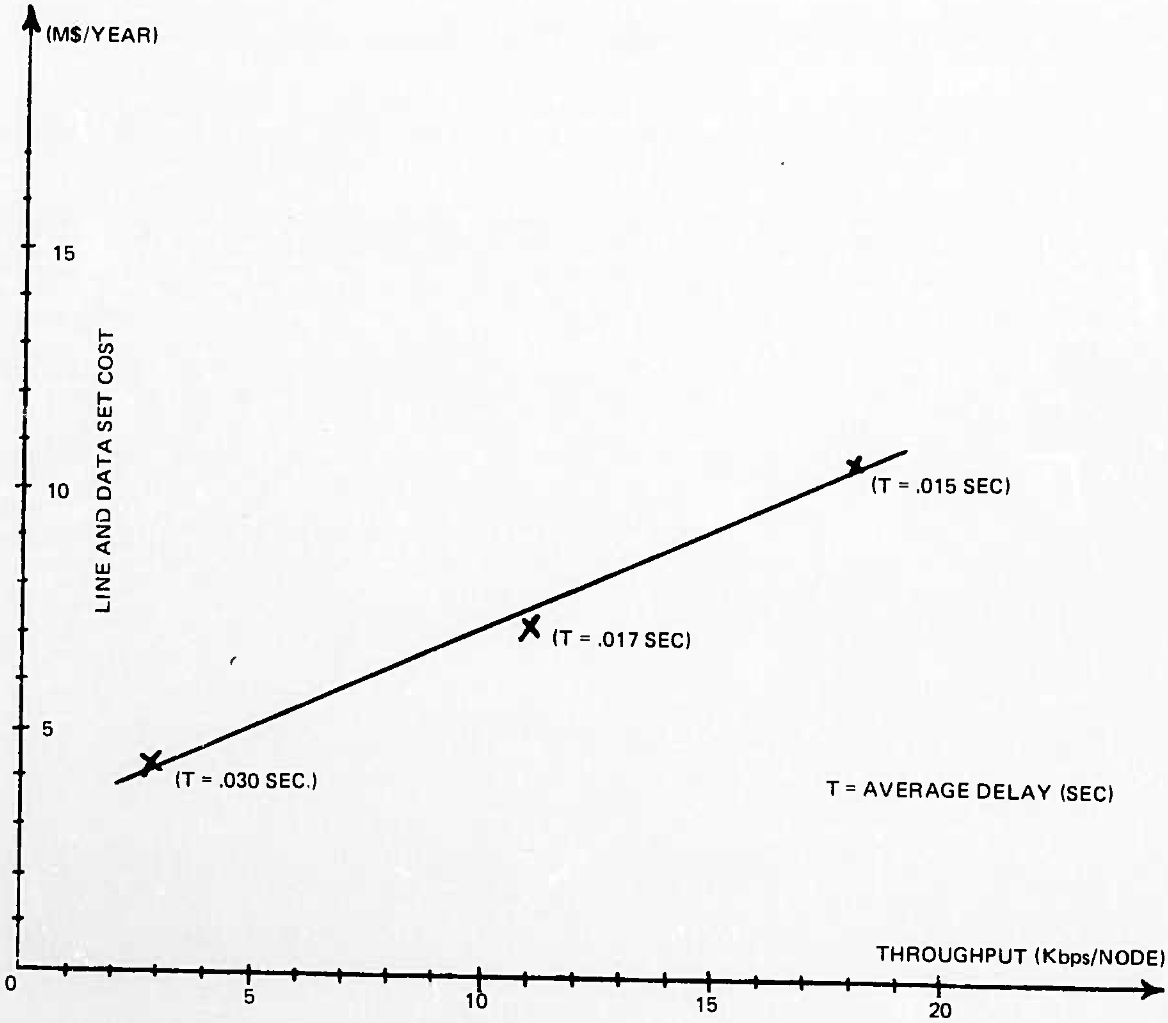


Figure 4.6: National Network

c. Highest cost net:
   12 x DDP-316 IMPs, cost = 180K$/year
   6 x HSMIMPs, cost = 450K$/year
   Total cost = 630K$/year

Figure 4.8 shows the results for the *local* net. Both 3-connected and loop configurations were analyzed. Various capacity assignments, leading to different solutions, were considered. Average delay T in the local nets is much higher than in the national and regional nets, because of the extensive use of 9.6 Kbps and 19.2 Kbps channels, especially in the low cost, low throughput configurations. The delay can be reduced by reducing the traffic load, as shown in Figure 4.8. The local network does not require, in general, redundant processors; the message processor cost is given by:

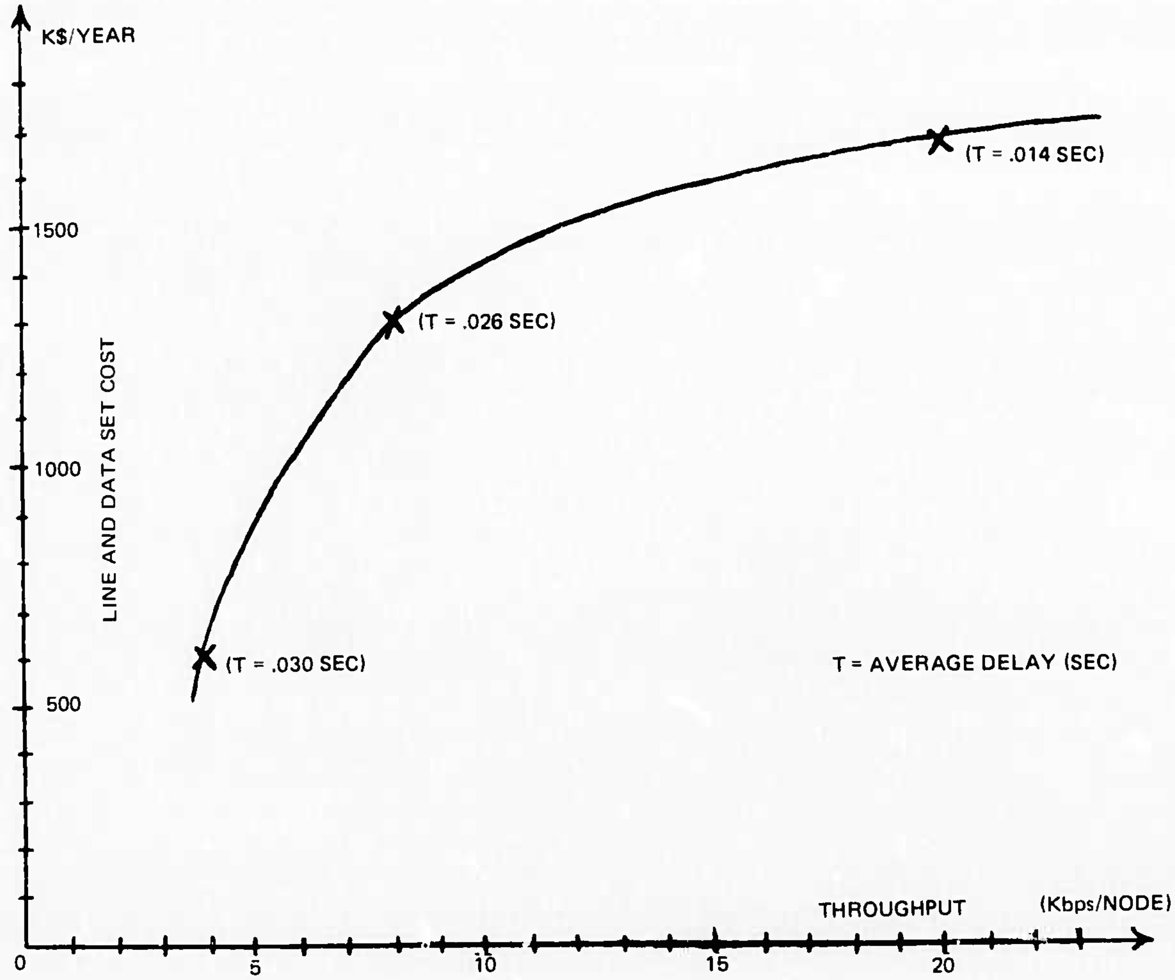


Figure 4.7: Regional Network (Texas)

Local network:
9 x DDP-316 IMPs, cost = 135K$/year

The results for the *global* net are obtained as follows:

a. For each throughput level, the lowest cost national, regional and local solutions that can accommodate such a throughput are selected.

b. The total cost D_t is given by:

$$D_t = D_n + 10D_r + 100 \times D_\ell$$

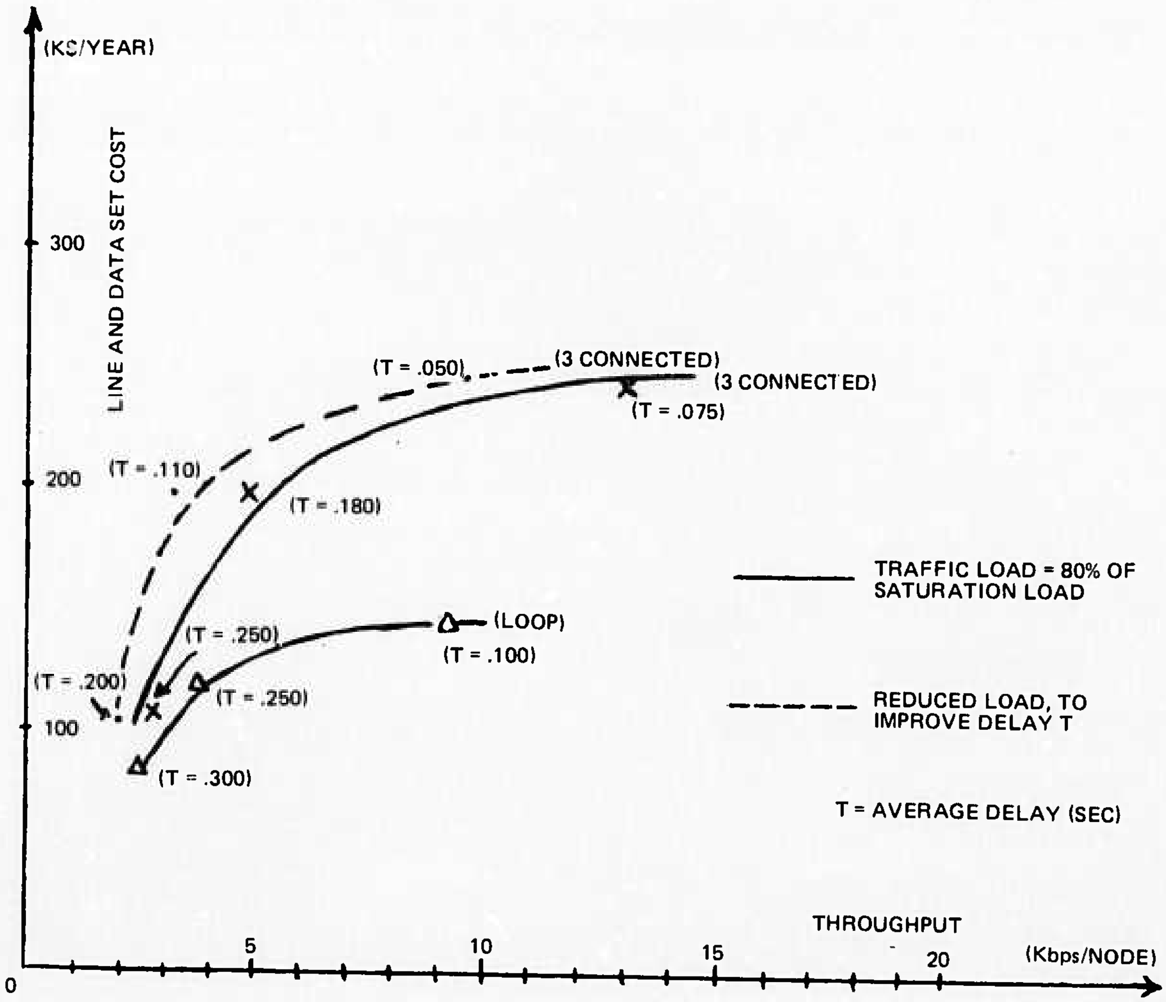


Figure 4.8: Local Network (Section of Texas)

where D_n = national net cost

D_r = regional net cost

D_ℓ = local net cost

c. The total average delay T_t suffered by a packet traveling from source to destination is typically given by:

$$T_t = T_n + 2T_r + 2T_\ell$$

where T_n = national network delay

T_r = regional network delay

T_ℓ = local network delay

Figure 4.9 shows channel cost and delay of the 1000 node net for both 3-connected and loop local net configurations while Figure 4.10 shows total communication cost.

The diagram in Figure 4.11 displays line and modem cost per node versus network size, for two different values of throughput. The shadowed area represents the cost of networks with local connectivity ranging from 2 to 3 for the 1000 node design. The cost for N=1000 seems to be slightly higher than the trend displayed for N up to 200. It should be remembered, however, that:

a. The cost estimate for N=1000 is not precise

b. The cost for N $\leqslant$ 200 was minimized using link exchange procedures [18], while the cost for N=1000 is the cost of feasible but unoptimized network

Thus, we can expect that optimized network cost for N=1000 would be lower and could follow closely the trend established for N up to 200. The upper bound for the N=1000 curve represents 3-connected local topology as well as 3-connected regional and national

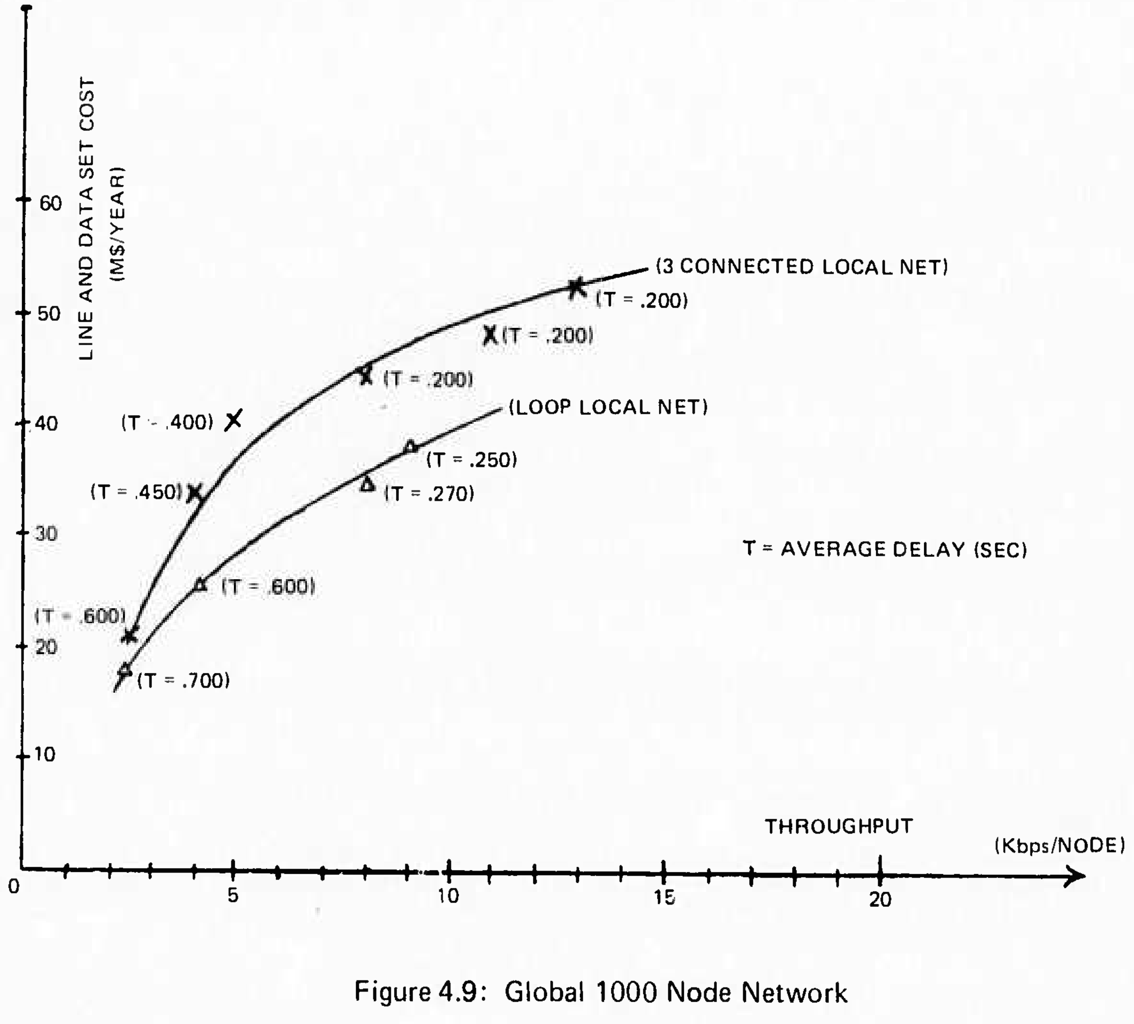


Figure 4.9: Global 1000 Node Network

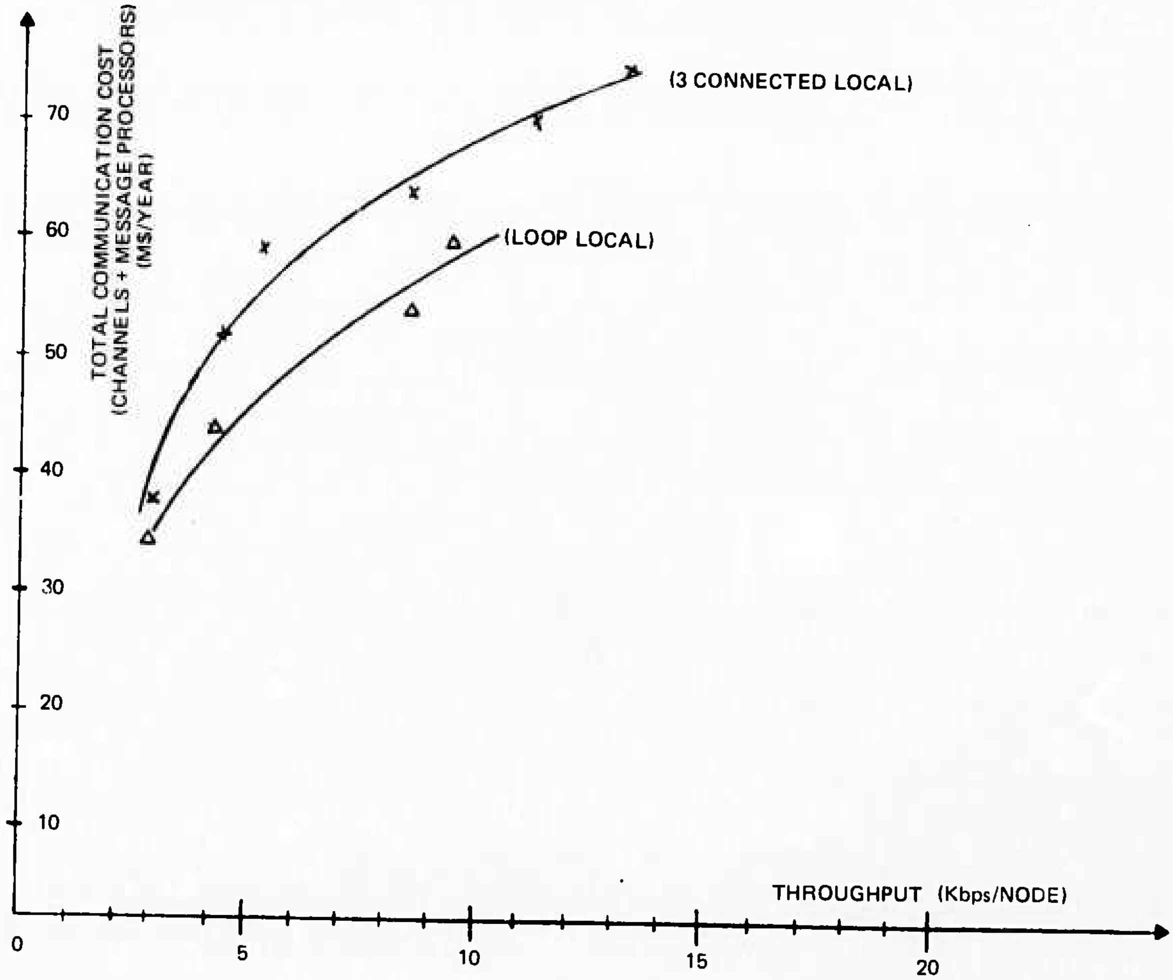


Figure 4.10: Global Node Network

topologies. Since this network can have very high reliability *without* local backup of nodes, we need never spend more than this bound for the 1000 node network. The region between the 2-connected and 3-connected points represents costs that might be achieved using optimization on each local network (with reliability as a constraint). Further economies would require the restructuring of the overall network hierarchy and partitioning achemes. We have noted that this optimization problem is extremely difficult.

4.6 Implications for Future Research

The results summarized here establish the feasibility, in terms of design techniques, cost, delay and reliability, of very large packet switched networks. Future steps in the research will be:

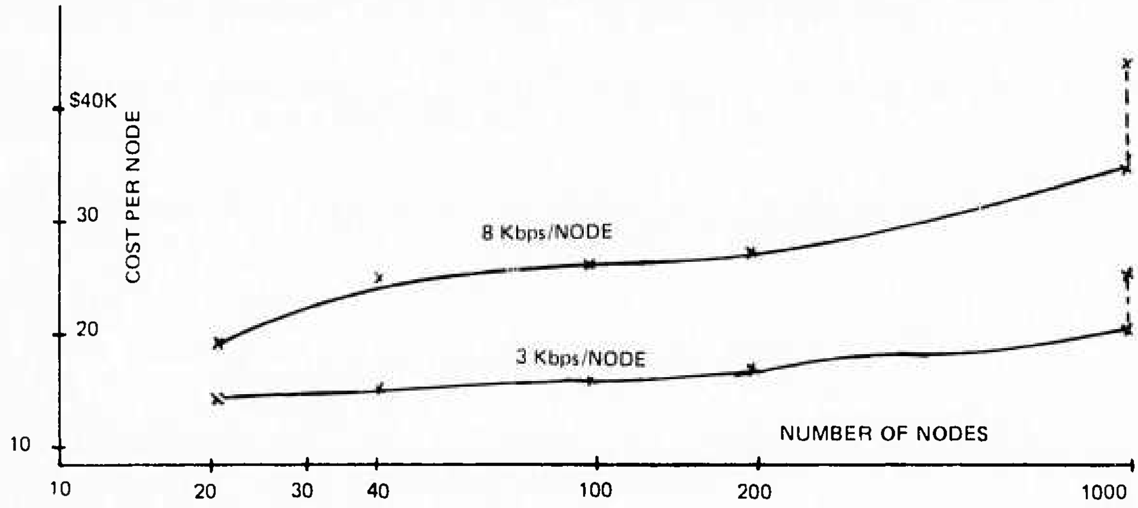


Figure 4.11: Network Size vs. Costs For Two Traffic Requirements

- Optimization of network design
- Performance evaluation
- Routing and flow control
- Use of different communication techniques at different hierarchical levels.

Some of the open areas are elaborated in the following.

4.6.1 Optimal Design

The design of hierarchical network requires: selection of number of hierarchical levels and of number of "nodes" for each level; determination of node partitions (on the basis of geographical distance, node requirements, etc.); separate minimum cost design for each partition and hierarchical level; combination of the partial designs into the global design. Low cost designs can be obtained with an iterative procedure, in which an initial configuration is successively improved, by properly modifying node partitions, local topologies, interconnections between different hierarchical levels, etc., until no more improvement is possible. One of the bottlenecks of the procedures is the local minimum cost network design, which must satisfy both traffic and reliability constraints. Present optimization techniques are inadequate and faster, and more efficient methods must be developed.

4.6.2 Performance Evaluation

The exact evaluation of throughput, delay and reliability for a 1000 node network requires a prohibitive computation time and memory space if performed with the present methods.

This is not so critical for the network design since approximate expressions of throughput, delay and reliability are probably sufficient. For the final configuration, however, a more precise performance evaluation is desirable, and therefore, new and efficient methods of large network analysis must be developed.

4.6.3 Routing and Flow Control

The traffic within each subnetwork can be routed and controlled with the present ARPANET techniques. However, proper modifications must be introduced to direct the traffic to external destinations. In addition, a multilevel flow control procedure could be implemented to obtain more efficient control of the traffic load in each hierarchical level.

4.6.4 Hybrid Communication Implementations

The hierarchical structure allows within certain limits, the use of different system implementations at different hierarchical levels. This feature can be exploited to obtain a more economical and efficient system. Possible configurations might include: broadcast "ALOHA type" radio techniques at the local level; packet switching techniques at the regional level; satellite broadcast techniques at the national level. It is of interest to investigate feasibility and economics of such hybrid implementations.

Chapter 5
RANDOM ACCESS PACKET TRANSMISSION

The most expensive part of a hierarchical communication system is often the lowest level since it contains the most elements. Local distribution and collection of data is usually characterized by very low utilization of facilities. Random access packet techniques provide promising schemes for satellite cable, and local radio communication. Hence, a basic understanding of these mechanisms is useful to the following chapters.

The exposition is aided here by one of those happy situations where the precise derivation is its own most intuitive explanation. We therefore present the derivation of the capacity of a random accessed channel, as originally devised for the ALOHA radio system [1]. Random access packet transmission is similar to time division multiple access with the following difference:

- Messages are broken into fixed size packets.

- Each transmitter can initiate a message at any time without reserving slots or requesting channel access.

Let τ be the duration of a packet and assume there are k active users. The overlap of two packets from different stations is illustrated in Figure 5.1. Assume that when an overlap occurs neither packet is received without error and both packets are therefore retransmitted. We also assume only full packets are transmitted.

Let those packets transmitting a given message from a station for the first time be message packets and let those packets transmitted as repetitions of a message be called repetitions. Let λ be the average rate of occurrence of message packets from a single active user and assume this rate is identical from user to user. Then the random point process consisting of the starting times of message packets from all the active users has an average rate of occurrence of:

$$r = k\lambda$$

where r is the average number of message packets per unit time from the k active users. Then, if we were able to pack the messages into the available channel space perfectly with absolutely no space between messages, we have:

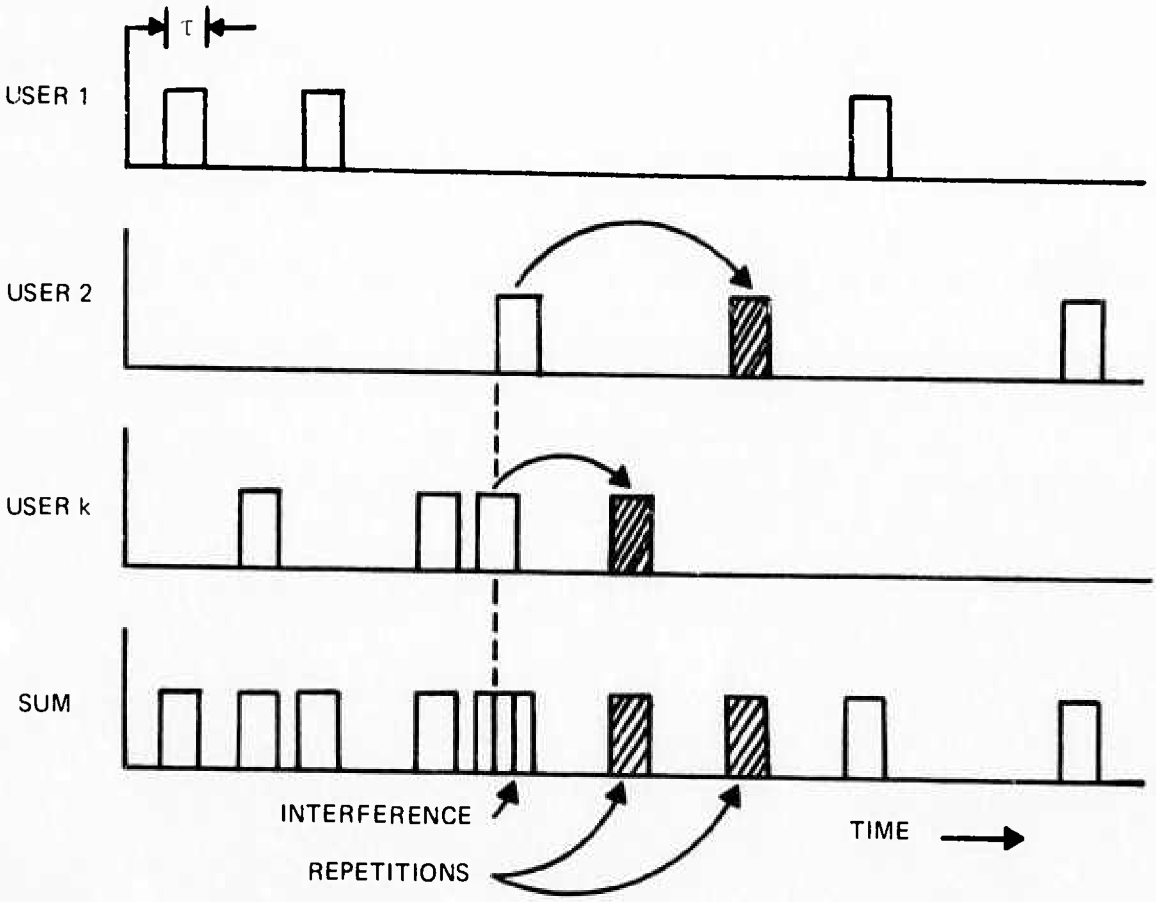


Figure 5.1: Random Access Packet Multiplexing

$$r\lambda = 1$$

Accordingly, we refer to $r\lambda$ as the channel utilization. We will determine the maximum value of the channel utilization, and thus the maximum value of k, which this random access data communication channel can support.

Define R as the average number of message packets plus retransmissions per unit time from the k active users. Define $R\tau$ as the *channel traffic* since this quantity represents the average number of message packets plus retransmissions per unit time multiplied by the duration of each packet or retransmission. We now calculate $R\tau$ as a function of the channel utilization, $r\tau$.

Assume the interarrival times of the point process defined by the start times of all the message packets plus retransmissions are independent and exponential. If the retransmission delay is large compared to τ , and the number of retransmissions is not too large, this assumption will be reasonably close to the true distribution. Under the exponential assumption, the probability that there will be no events (starts of message packets or retransmissions) in a time interval T is exp(-RT).

Two packets overlap if there exists at least one other start point τ or less seconds before or τ or less seconds after the start of a given packet. Hence, the probability that a given message packet or retransmission will be repeated because of interference with another message packet is:

$$[1 - \exp(-2R\tau)]$$

Thus, the average number of retransmissions per unit time is:

$$R[1 - \exp(-2R\tau)]$$

Therefore,

$$R = r + R[1 - \exp(-2R\tau)]$$

$$r = R\tau e^{-2R\tau}$$

The plot of $R\tau$ versus $r\tau$ is given in Figure 5.2.

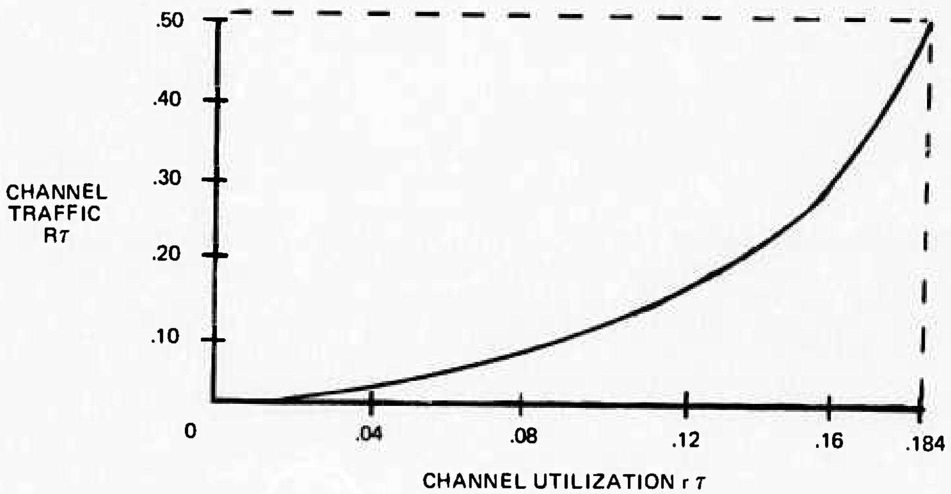


Figure 5.2: Channel Utilization vs. Channel Traffic

The channel utilization reaches a maximum value of $\frac{1}{2e} = 0.184$. For this value of $r\tau$ the channel traffic is equal to 0.5. The traffic on the channel becomes unstable at $r\tau = \frac{1}{2e}$ and the average number of retransmissions becomes unbounded. Thus, we may speak of this value of the channel utilization as the *capacity* of this random access data channel. Because of the random access feature, the channel capacity is reduced to roughly one sixth of its value if we were able to fill the channel with a continuous stream of uninterrupted data. To obtain the maximum number of interactive users the system can support, we get:

$$r\tau - k\lambda\tau = \frac{1}{2e}$$

Solving for the maximum number of active users, we have:

$$k_{max} = (2e\lambda\tau)^{-1}$$

A modification of the system called a slotted system allows message origination only at fixed intervals. It introduces some synchronization problems, but raises k_{max} to $(e\lambda\tau)^{-1}$. It is most important to realize that k_{max} is the number of users who can use the communications channel simultaneously. In contrast to the usual frequency or time multiplexing methods, while a user is not active, he consumes no channel capacity so that the total number of users of the system can be considerably greater than k_{max}.

Chapter 6
UPGRADING A TERRESTRIAL NETWORK USING SATELLITE LINKS

To illustrate the complexity of the decisions involved in upgrading communications, we now give one simplified example developed in Semiannual Report No. 2 of the introduction of a satellite into an ARPANET. The costs of the satellite facilities are roughly comparable to those proposed by a number of satellite companies with proposals already approved by the FCC.

The satellite facilities include: satellite channel; ground stations; Satellite Interface Message Processor (SIMP); and line connections from SIMP to station, or from IMP to station.

The following costs are assumed:

a. Satellite Segment:

Bandwidth (Kbps)	Costs ($/Mo.)
50 full duplex	2,500
230 full duplex	5,500
1,500	8,000

b. Local Loop (Station to SIMP, or Station to Central Office):

Bandwidth (Kbps)	Cost ($/Mo.)
50 full duplex	1,000
230 full duplex	1,300

c. SIMP. Two types of SIMP's are assumed. The regular SIMP has bandwidth greater than 1,500 Kbps and cost of 5,500 $/mo. This SIMP corresponds to the high speed version of IMP presently under development of BBN. It can support a combination of land traffic rates L and satellite traffic rate S such that:

$$L + 3S \leqslant 1{,}500$$

A small SIMP, with a bandwidth of 600 Kbps, and cost of 1,400 $/mo., is structurally similar to the H-316 IMP, and is presently being developed by BBN. The throughput constraint is :

$$L + 3S \leqslant 600$$

The network without satellite links used as a basis of comparison is a recent 43 node ARPANET configuration shown in Figure 6.1. We then consider one upgrade (see Figure 6.2) using only terrestrial links as well as two upgrades (see Figures 6.3 and 6.4) of the network using satellite links as well. In both satellite designs, five ground stations in San Francisco, Los Angeles, Washington, D.C., New York and Chicago are available for satellite access. We include the possibility of capacity reductions of terrestrial links from 50 Kbps to 19.2 and 9.6 Kbps. In the first satellite design, we use point-to-point access which divides the satellite channel bandwidth into subchannels, each corresponding to a full duplex point-to-point connection between given ground stations. In the second satellite design, we use the slotted random access packet mode described in the last Section. Computed for each network configuration is: total cost; terrestrial costs of all terrestrial links; satellite cost; cost of SIMP (if applicable); connection from SIMP to station or from IMP to station, and satellite bandwidth; total throughput; traffic on satellite and satellite channel delay. The results are in Table 6.1.

Several observations can be made from this simple example. Because of the high cost of network to ground station connections, satellite links become attractive only for throughput levels which are about 50% higher than the ARPANET configuration. Furthermore, even to achieve these efficiencies, changes in network topology and reduction of some link

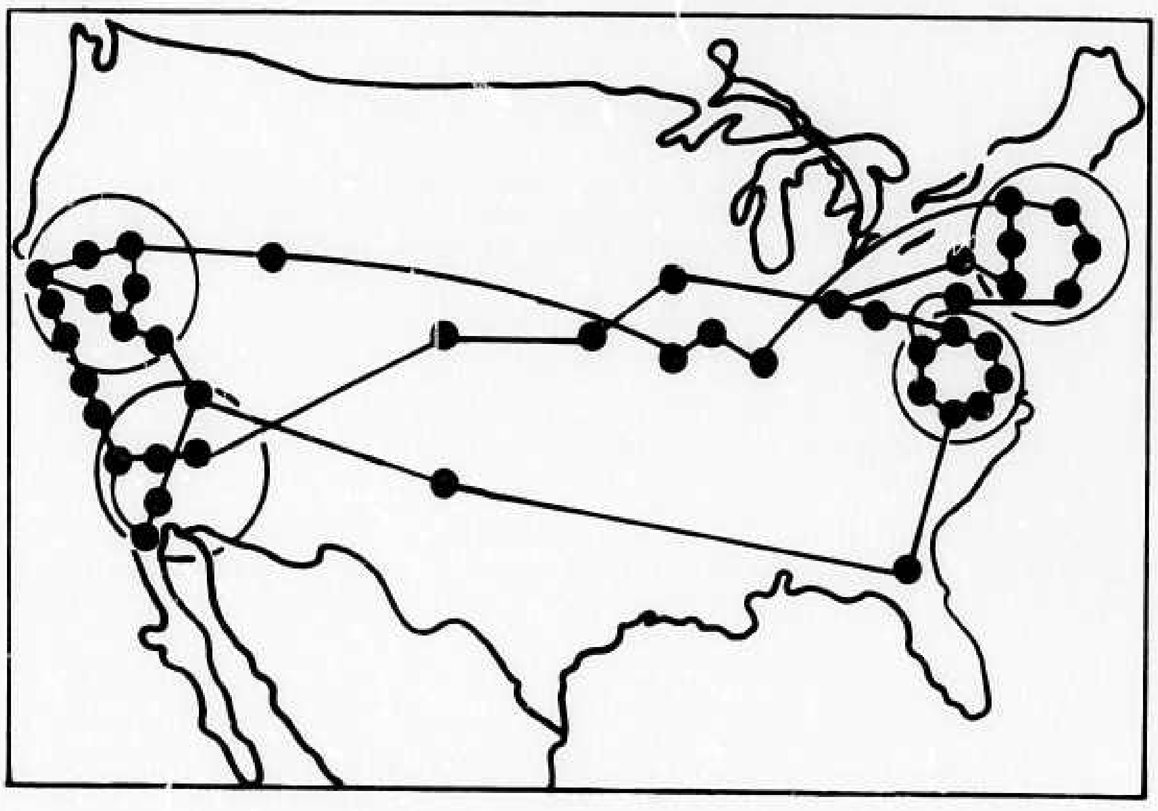


Figure 6.1: Present ARPANET Configuration (October, 1973)

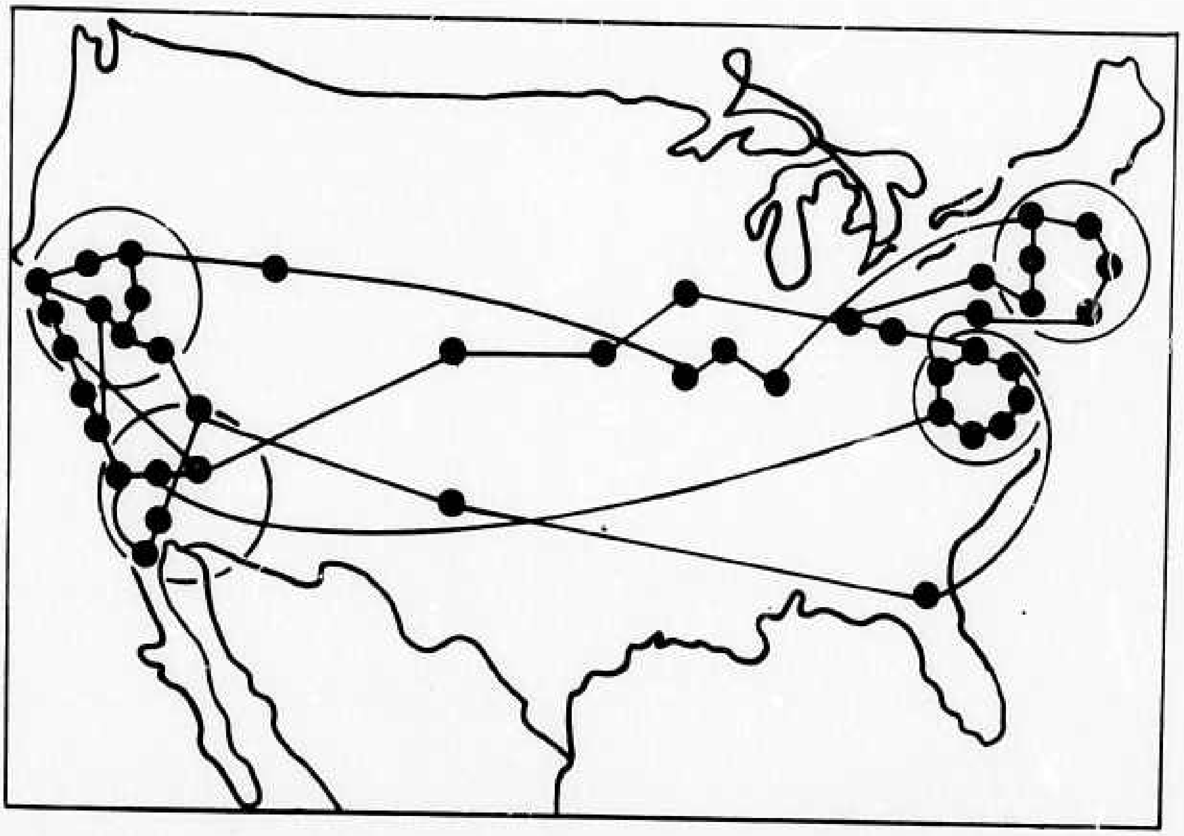


Figure 6.2: Upgraded 43-Node Configuration

capacities to 19.2 Kbps and 9.6 Kbps are required. Examination of the designs also indicates that the proper location of ground stations is important. For example, the location of a ground station in Chicago allows the reduction of channel capacity on cross country connections.

The comparison of cost-throughput trends between implementations with and without satellite, when network throughput is increased, shows that satellite implementations can provide higher throughput at a lower cost, especially if the terrestrial network is reoptimized; but the savings are by no means guaranteed by only routine introduction of satellite links. Many tradeoffs are involved and careful optimization is required.

Furthermore, it is clear that many other factors disregarded in this simplified analysis must be taken into consideration before general cost-performance trends are evident. In particular, the evaluation of point-to-point satellite link cost assumed that the standard IMP software can support satellite rates up to 230 Kbps. There are indications, however, that such a high rate will require modifications of the IMP hardware and software, and therefore will raise the cost of point-to-point links to the same levels as those of random access.

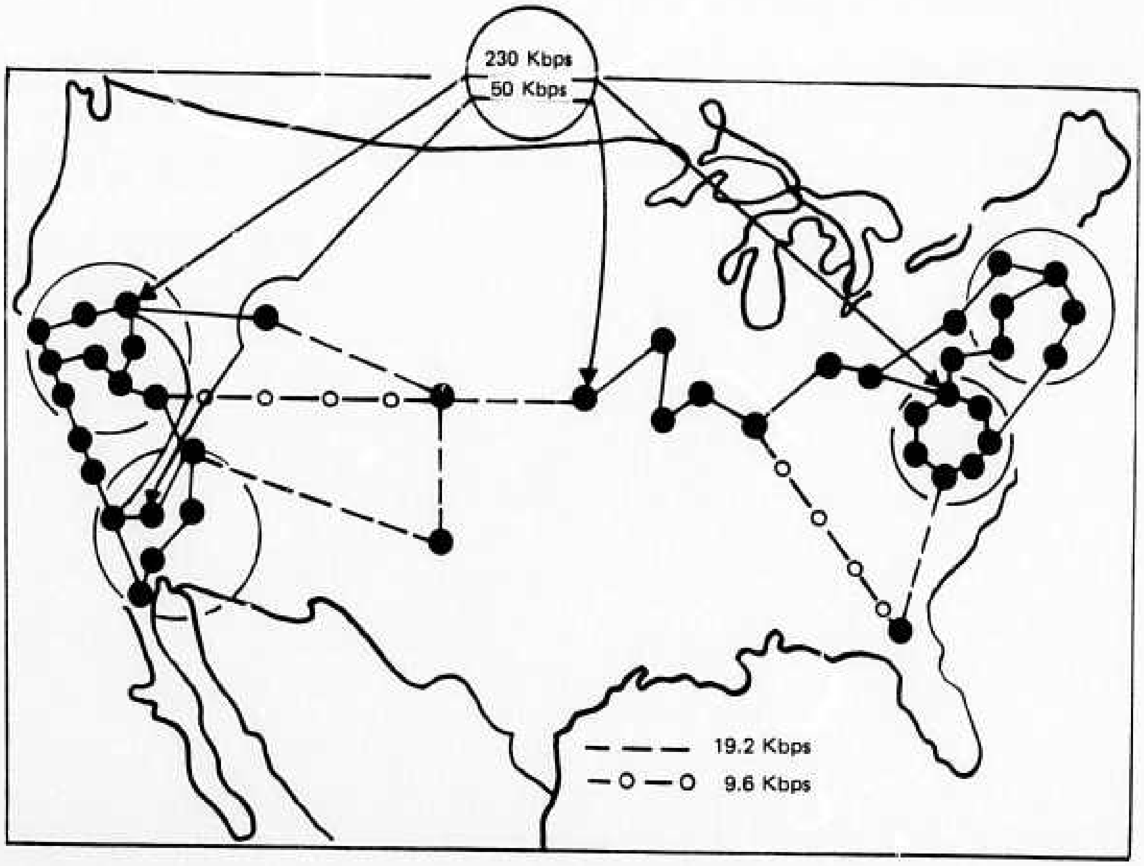


Figure 6.3: 2 Point-To-Point Links

As for satellite bandwidth efficiency, it must be mentioned that, with additional software cost, reservation techniques for multiple access can be implemented on the SIMP; and such techniques can theoretically increase effective satellite bandwidth up to full utilization. Furthermore, multiple access allocates satellite bandwidth dynamically, according to traffic pattern changes, and if needed, allows any two stations to use the full channel; while point-to-point access corresponds to a rigid bandwidth allocation between pairs of stations.

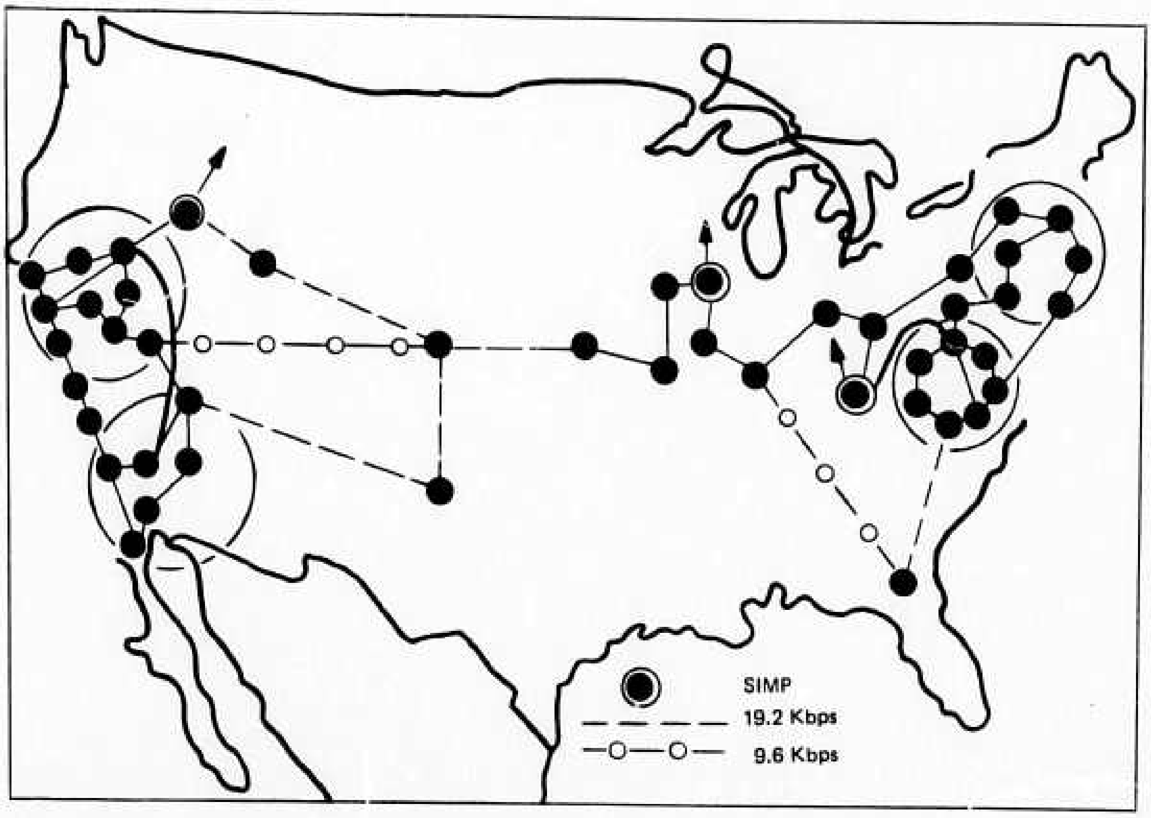


Figure 6.4: 3 Regular SIMP's

Table 6.1: Upgrading of Packet Switched Terrestrial Network Using Satellite Links

Network Configuration	Satellite Delay (sec)	Satellite Traffic (Kbps)	Cost K$/mo. Terrestrial	Cost K$/mo. Satellite	Cost K$/mo. Total	Throughput Kbps	Total Cost Throughput $/bit
Present 43 node configuration (Figure 6.1)	.....	......	93	....	93	447	.208
Upgraded 43 node configuration (Figure 6.2)	.....	......	112.9	....	112.9	635	.178
Two point to point links (Figure 6.3)	.27	330	75	16.9	91.9	654	.141
3 regular SIMPS at San Francisco, Chicago and Washington, D.C. (Figure 6.4)	.5	393	79	29.3	108.3	686	.158

Chapter 7
MULTIDROPPED LINE NETWORKS FOR LOCAL ACCESS

7.1 Introduction

The interconnection of different time-sharing computers through a sophisticated communications subnet in the ARPANET gives terminal users access to a variety of time-sharing resources. Initially, ARPANET development was directed toward computer-computer communications and user protocols. Originally, only terminals connected directly to a computer in the network had access to the network. The successful completion of this initial phase led to a desire to complement resource development with increased user access. Many TIP's have already been installed and are currently in use, connecting users with a terminal, but with no local Host computer, to the network.

The use of the ARPANET approach within the Defense Department would involve hundreds of Hosts accessed by tens of thousands of low speed terminals. Effective, economical terminal access to the ARPANET, and to similar networks, will depend on continued development of such facilities as TIP's as well as on complementary development of techniques for cost-effective utilization of these facilities.

There are several ways to provide terminal access into the network. In particular, multidrop lines for connecting terminals to access ports, ring networks, CATV Systems and packet radio techniques all provide potential low cost network access methods. It is necessary to investigate all of these schemes to evaluate the merits of each and to determine the conditions under which each may be preferable. It is not unreasonable to anticipate that many approaches may be applicable within the same network. In this section, we consider multidrop lines. Other local access methods are considered in the following sections.

In NAC's semiannual reports, algorithms are described for the multidrop line-layout and TIP location problems. These algorithms consider not only the line layout, but also the number, location, and characteristics of the ports into the network. Models were developed to estimate the cost of connecting terminals to the network through the use of TIP's and multidrop lines. The estimates cover a wide range of terminal numbers and traffic conditions and serve as a basis for comparison with other access approaches and as a measure of the effectiveness of new design tools.

7.2 Network Modeling for Terminal Access

The process of investigating and developing approaches to the design of networks for terminal access requires, as a first step, the construction of an appropriate model. In this study, a primary goal is to determine the tradeoffs and parametric dependencies present in various approaches to terminal access. To compare these approaches, it is necessary to have a common data base to which each approach can be applied. Such a base has been constructed in the form of a model for the number and geographic distribution of terminals, for the terminal and terminal user, and for the multidrop communication lines. It is also necessary to model those components of a design peculiar to the particular approach considered. In this report we present a model for the TIP system.

7.2.1 Population

The cost of terminal access will depend on a variety of factors, including the number of terminals to be connected and their geographic distribution. To determine the parametric dependence of cost on the number of terminals, populations of from 100 to 2000 terminals were considered. The figure of 100 reflects the anticipated near-future requests for terminals. The figure of 2,000 reflects an order of magnitude estimate of the number of terminals that can be expected to be served by a fully developed network. (The above consider only terminals without a local Host.)

To have a meaningful data base, it is necessary to geographically distribute the terminals in a sensible manner. Terminals were located on the basis of population density because of the success of this approach in previous NAC investigations. A rectangular region was determined for each city, or collection of cities, to reflect the feasibility of the region to support a population segment with access to urban facilities. Thus, consideration was given to natural geographical boundaries, such as mountains, lakes, and coast lines, to major roads in the area, to the number of nearby smaller communities, and to the natural pattern of urbanization between relatively close major population centers. Using this approach, 123 regions were defined, with varying sizes of approximately 70 square miles.

Once a number of terminals has been allocated to a region in proportion to population, the geographic positions of the terminals within the region are uniformly randomly distributed. With a large number of terminals, it is reasonable to anticipate that some may be located at points with no discernible geographic significance; therefore, a fraction a of the terminals were located at random in a large geographic segment: east of Denver, west of Pittsburgh, north of Austin, and south of Milwaukee. The fraction a was selected on a sliding scale as shown in Table 7.1.

7.2.2 Terminal-Terminal User

Even though network resources in the ARPANET have been extended far beyond traditional time-sharing, the interactive user retains a significant role in network usage, and the

Table 7.1: Terminal Population

Number of Terminals	In Regions	Random
100	95%	5%
200	95%	5%
500	90%	10%
1,000	90%	10%
2,000	85%	15%

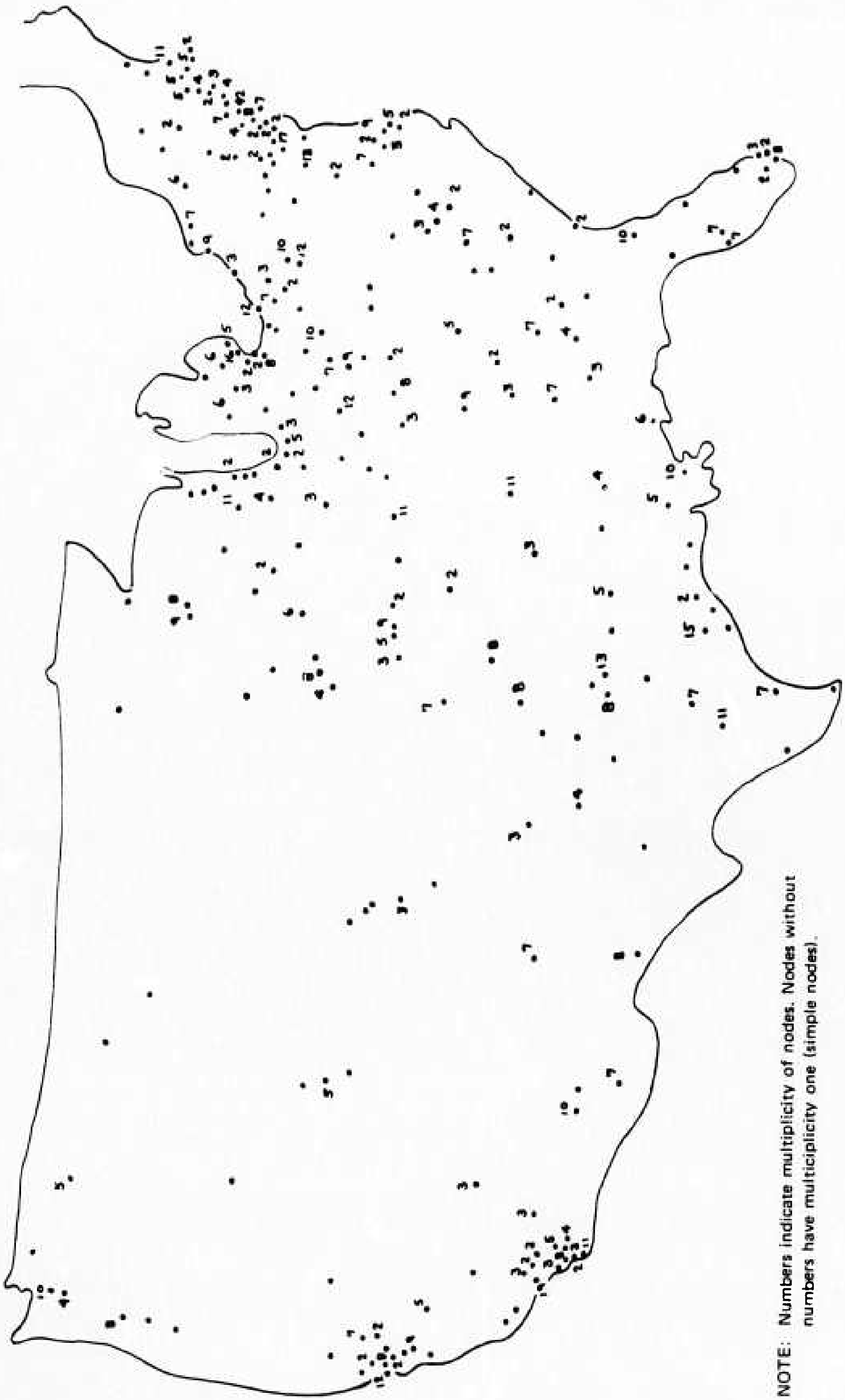
extension of accessibility to a terminal basis will give even greater significance to terminal-computer traffic. To effectively design and evaluate terminal access networks, it is necessary to model the terminal traffic. Two of the few definitive papers on time-sharing modeling from a communications perspective have been written by Jackson and Stubbs [33], and Fuchs and Jackson [22]. The following traffic characteristics of a terminal during a period of use are based on their results for time-sharing systems used in scientific applications, and extended in consideration of advances in terminal technology, higher speed lines being used, and more sophisticated time-sharing users and programs.

	Average Message Length	Minimum Average Traffic	Maximum Average Traffic
User Input	12.1 characters	.1 characters/second	1 characters/second
Computer Response	52.8 characters	1.0 characters/second	10 characters/second

In this study, traffic level was varied, with a range of variation from the minimum to the maximum values indicated above. The minimum average traffic level reflects the results of the noted study for scientific applications using low speed facilities and ordinary time-sharing programs. The maximum average traffic level reflects an extension of these results in consideration of "smart-fast" terminals, higher speed communication facilities, more advanded time-sharing programs, and more sophisticated users. For comparison purposes, all network designs were based on busy hour conditions of all terminals being active.

7.2.3 Communication Facilities

The current ports for access by terminals to the ARPANET (TIP's) may connect terminals directly, or remotely through modems and phone lines. In this study, a large number of terminals serving interactive users are considered, and to economically connect all the terminals, multidrop lines were used. The multidrop communications facility will be assumed to be a standard voice-grade line as described below.



NOTE: Numbers indicate multiplicity of nodes. Nodes without numbers have multiplicity one (simple nodes).

Figure 7.1: Locations of the 1000 Nodes

7.2.3.1 Multidrop Line

Capacity (full duplex)	1200 bps
Cost	\$.50/mile + \$40/drop

The monthly cost is based on the Government rate of \$.42/mile plus 20% for non-direct routing. It should be noted that in this model the number of drops on a line is restricted only by the traffic constraint. In reality, the number is often additionally restricted by telephone company practices. The effect of a more severe restriction is easily seen by simply assuming a correspondingly higher traffic level.

7.2.3.2 TIP

The approach considered will be a TIP serving as the root of a centralized network of terminals. In this section we note the significant features of the TIP. The TIP, as described by Ornstein et. al [44], is characterized in Figure 7.2. Its characteristics indicate:

a. The TIP has 63 terminal I/O slots.

b. Each slot can handle direct terminal connections or connections via modems.

c. Asynchronous data rates handled by the TIP include 1200, 1800, and 2400 bps.

d. The TIP has a terminal program throughput of:
   - 100 Kbps one way traffic if messages are long ("many characters"), and
   - 5-10 Kbps if each terminal message is a single character.

e. The TIP uses 5% of its processing capacity to act as an IMP.

f. The TIP uses 10% of its processing capacity to field MLC interrupts.

g. The bandwidth capability of the TIP is summarized approximately by the formula:

$$P + H + 11T \leq 850$$

where P = total phone line traffic (Kbps)
H = total Host traffic (Kbps)
T = total terminal traffic (Kbps)

and full duplex units count twice baud rate, i.e., standard full duplex 50 Kbps phone line counts 100, and full duplex ASR-33 counts as 0.22. As noted, the TIP does not presently handle multidrop lines. Consequently, adjustments to its characteristics are required for the network model.

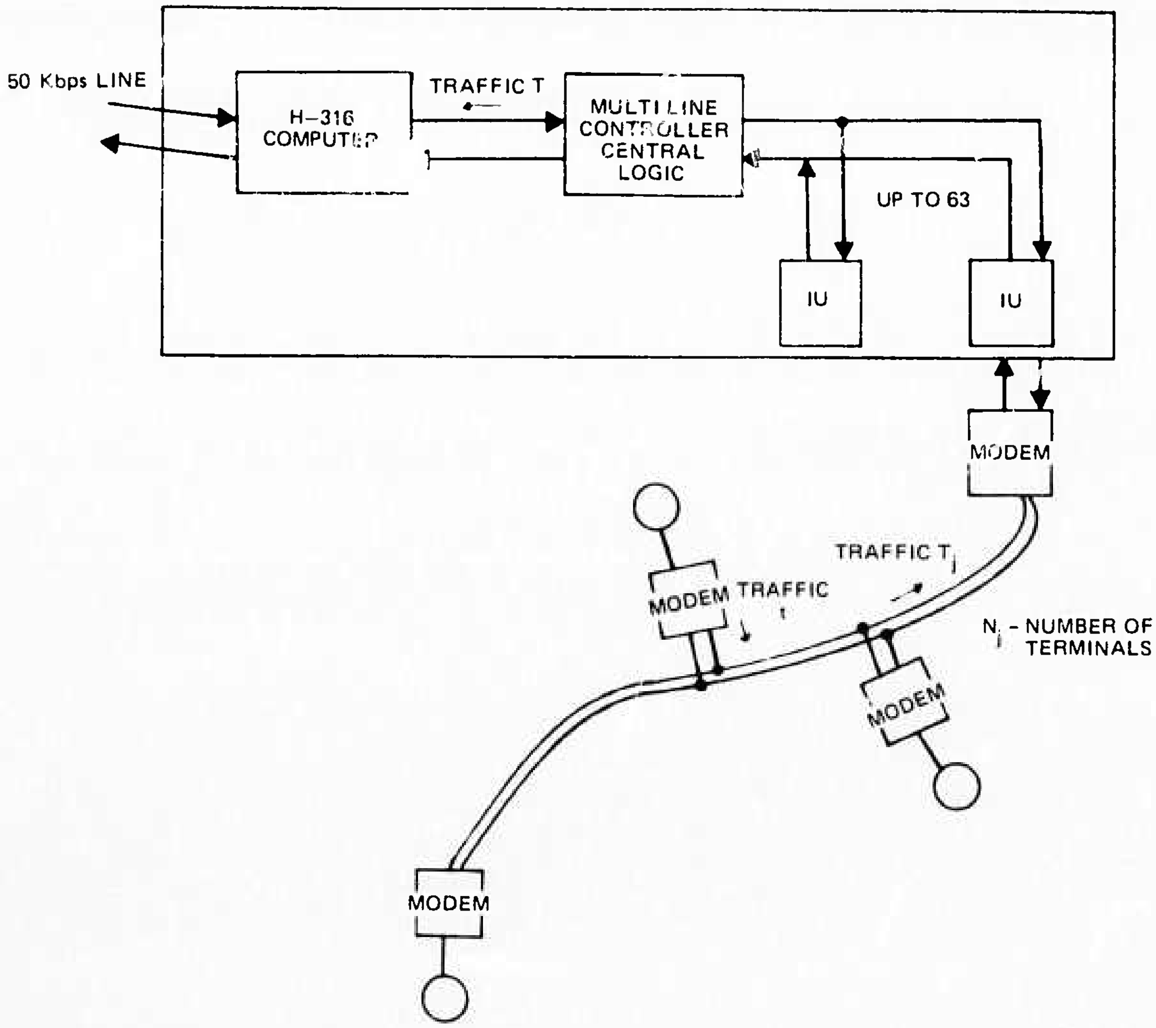


Figure 7.2: TIP Interfaces

The above noted characteristics reflect TIP capacity as a message concentrator for terminals. Additional consideration must be given to TIP cost, which includes estimated rental rate, the cost of its interconnect to the ARPANET, and the cost of the modems necessary to connect terminals. Therefore, the total cost is a function of the TIP's geographical relationship to the rest of the ARPANET, the topology of its interconnection, and the number of modems required for terminal connections. These costs will be determined as follows:

50 Kbps line (ARPANET interconnect) \$5/mile + \$425/end
(based on current ARPANET experience)

1200 bps line (terminal connection) \$17/modem
(current standard cost)

TIP rental $2500/month
(assumed TIP cost of $100,000 to be amortized over
5 years at 10% interest compounded quarterly)

7.3 Design Results

As noted, the currently designed TIP has no provision for the support of multidrop lines. Both hardware and software modifications may be necessary for the acceptable addition of this capability. Significant requirements are line protocol for the multidrop lines and more extensive file manipulation resulting from the larger number of terminals. The line protocol must permit line utilization of approximately 50%, a conservative figure based on the use of ordinary polling techniques for multidrop lines. With the previously described traffic range of 10 bps to 100 bps, this gives a possible range of 6 to 60 terminals on a line. The sixty-three possible connected lines allow a maximum demand of 37.8 Kbps to be placed on a TIP by the terminals. Using the maximum demand figure in the TIP bandwidth formula indicates that such a TIP would have sufficient additional bandwidth to support a Host and also be connected to the ARPANET in a manner consistent with current practices. However, the number of terminals a TIP handles in the maximum demand case (378 to 3780) is far beyond the current maximum configuration (63). This increase in number should be anticipated as causing considerable additional overhead for file manipulation. Furthermore, additional overhead may be anticipated due to the burden of a multidrop line protocol. Under these conditions, the maximum number of terminals that a TIP can handle is assumed to be 630, one order of magnitude greater than its current direct connection capacity. This gives a network model as below:

TIP
1) up to 63 line connections
2) up to 630 terminals

Lines
up to $\frac{600}{t}$ terminals/line
where t is the traffic/terminal in bps

Cost is estimated as a function of the number of terminals and their traffic level, subject to fixed TIP locations. In Table 7.2 below, costs are given for 100 terminal system at a traffic level of 100 bps each for different numbers of TIP's at different locations.

These results show that a higher number of TIP's yields lower line costs, but not necessarily a lower total cost. Consequently, the number of TIP's is varied until a local minimum is reached. Table 7.3 gives preliminary estimates of the cost of terminal connection as a function of the number of terminals and the level of traffic. Results are shown as points connected by straight line segments in Figure 7.3. The curves suggest that for low numbers of terminals, and thus, low numbers of TIP's, the line constraints and TIP locations have significant impact on cost. For large numbers of terminals, and thus, larger numbers of TIP's, costs are less sensitive to TIP placement and line constraints. Simplified illustrations of several of the network designs are given in Figures 7.4 through 7.7. Note

Table 7.2: Network Cost and TIP Location Relationship

# TIP's	Locations	100 Terminals (100 bps each) Monthly Line Costs	Monthly Line Costs And TIP Rental
1	Chicago	$14,007	$16,507
	Memphis	14,501	17,001
	New York	17,190	19,690
2	New York-Los Angeles	13,091	18,091
3	New York-Los Angeles-Chicago	11,375	18,875
	New York-Los Angeles-Chicago	11,302	18,802

Table 7.3: Preliminary Terminal—TIP Experiment Results

Number of Terminals	Traffic (bps)			
	10	20	50	100
100	$ 13,095	$ 13,231	$ 15,146	$ 17,607
200	18,906	19,875	23,373	31,818
500	36,050	39,138	49,208	56,099
1,000	66,775	72,893	83,886	94,189
2,000	119,570	125,085	144,759	165,817

that for low traffic (10 bps) it is cost effective to use as few TIP's as possible, while for high traffic (100 bps), savings are achieved by using more than the minimum number of TIP's. (With low traffic, many terminals can be chained together on one line to economically connect distant terminals to a TIP. With high traffic, only a few terminals can be placed on a line, and distant terminals result in several long, uneconomical lines.)

Since TIP's are relatively expensive when compared to conventional multiplexers, these simpler devices to achieve economy of scale will also be investigated as an alternative architecture.

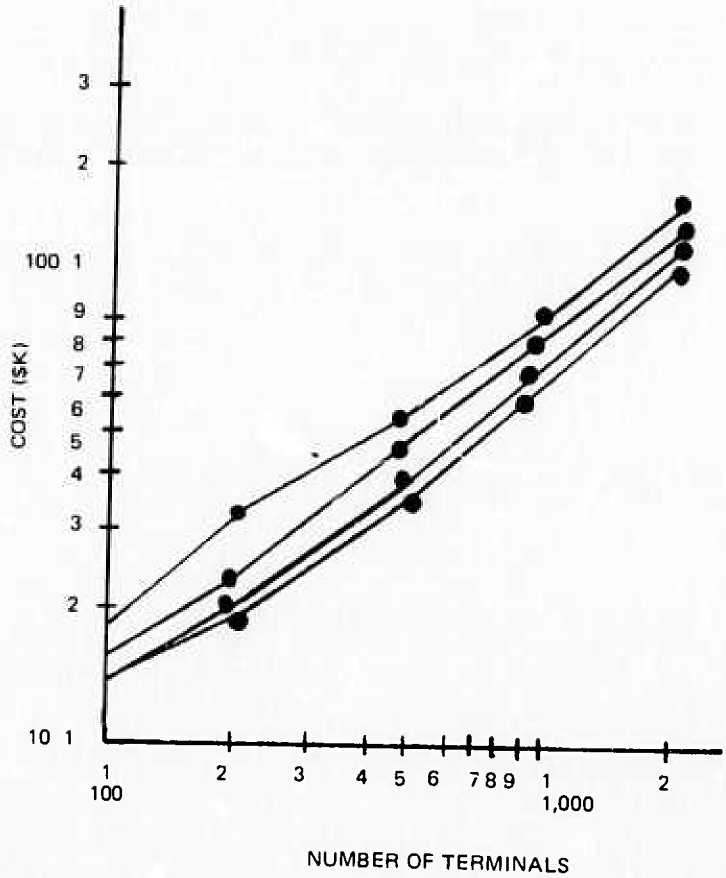


Figure 7.3: Preliminary Estimates of Terminal Connection Costs

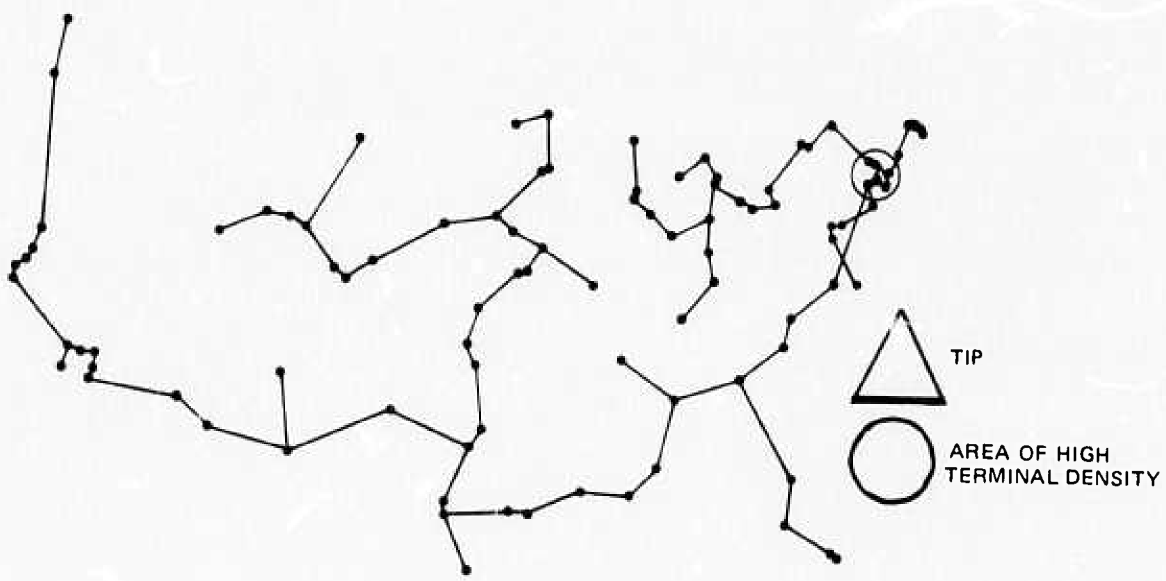


Figure 7.4: Network Design for 100 Nodes, 10bps Traffic, TIP in N.Y.C.

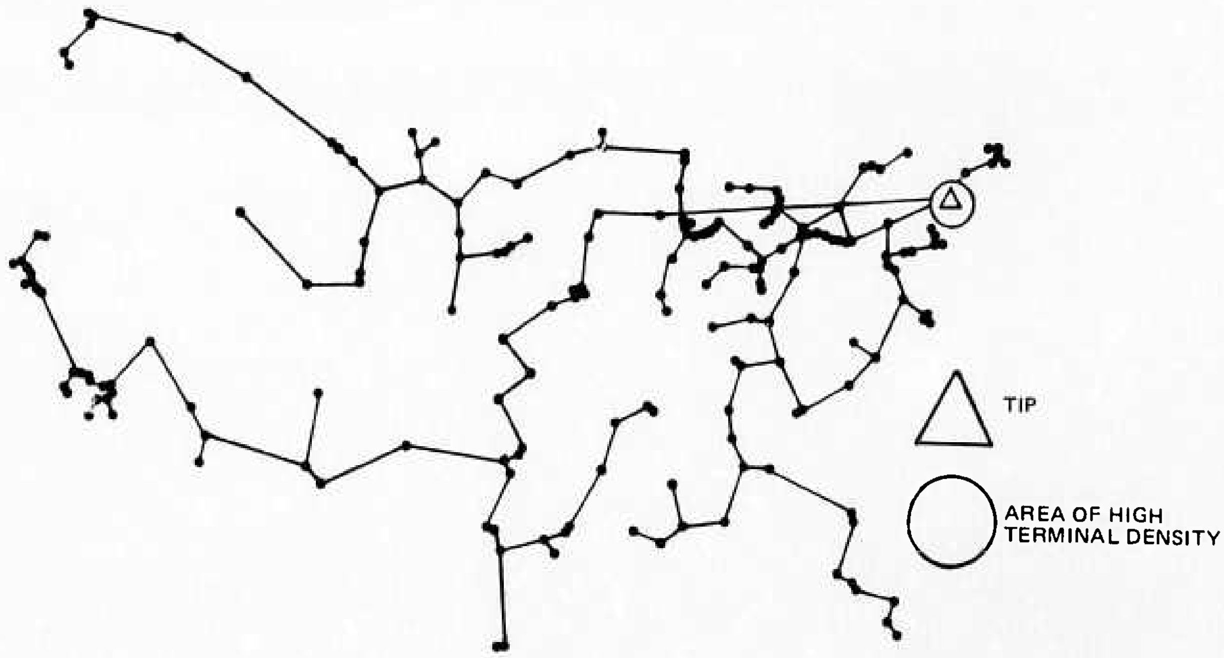


Figure 7.5: Network Design for 200 Nodes, 10bps Traffic, TIP in N.Y.C.

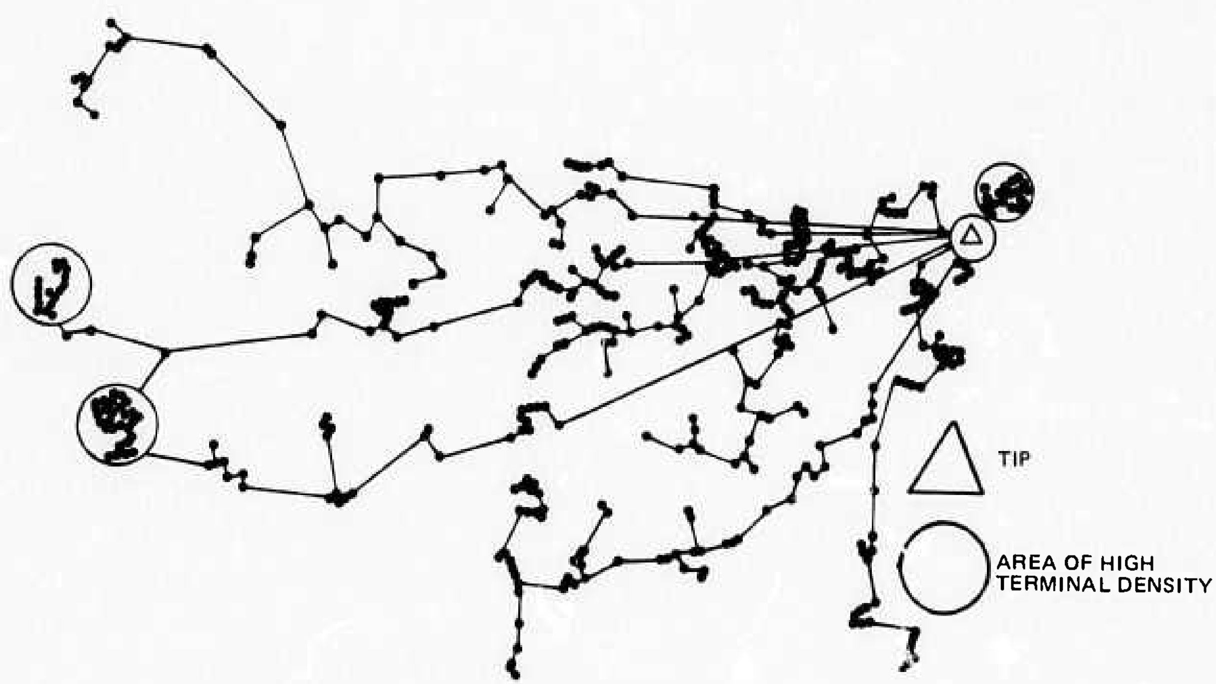


Figure 7.6: Network Design for 500 Nodes, 10bps Traffic, TIP in N.Y.C.

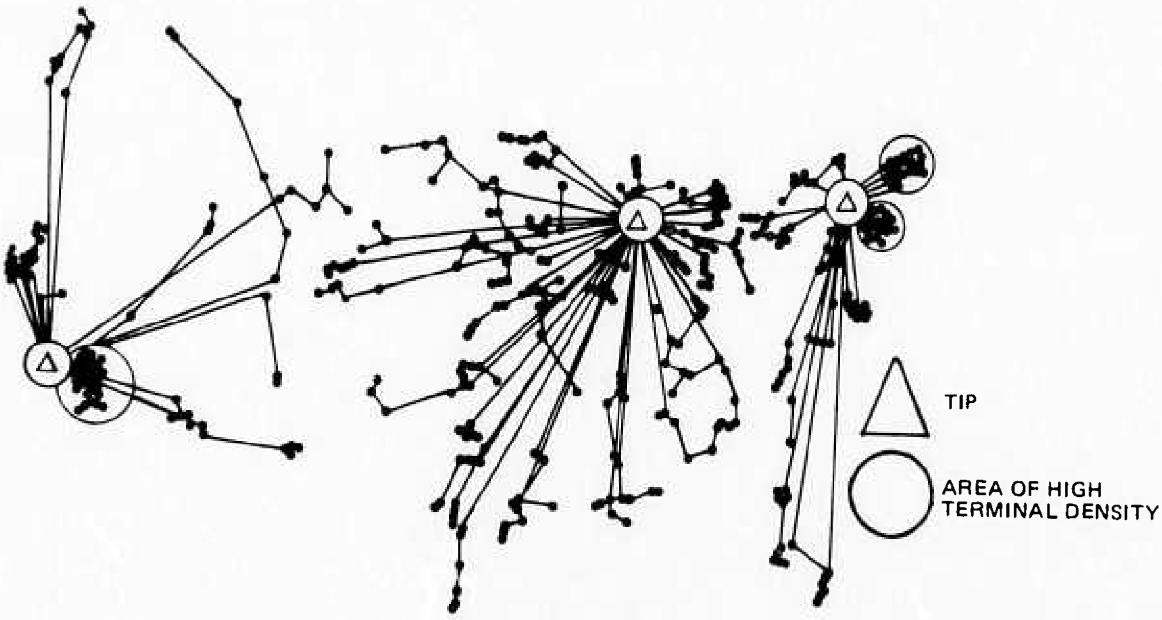


Figure 7.7: Network Design for 500 Nodes, 100bps Traffic, TIP in N.Y.C., Chicago, L.A.

Chapter 8
LOCAL ACCESS—A RING DESIGN EXAMPLE

For local transmission of signals from a nationwide interconnecting network, the user's technical problems are complex because many of the techniques are in the experimental stage. The problem is not just one of configuring facilities, but actually designing the channel. The classical technique of using multidrop lines with polling concentrators as described in Chapter 7 is available and in many cases the best strategy. But, new techniques such as the use of rings or random access multiplexing offer better prospects in many cases. However, neither of these are standard techniques and hence protocols and hardware are in a developmental stage. Furthermore, new physical links are becoming available. One of the most promising of these is the coaxial cable of cable television (CATV) systems.

To illustrate some of the complexities and surprises awaiting the designer of local systems, we present one example of a ring design. In Chapter 9, we present a detailed consideration of the use of CATV systems for local data transmission. In the remainder of this report, Chapters 10 through 15, we discuss the use of broadcast packet radio techniques for handling the local access problem.

Let us illustrate just one of the problems with a ring network, inflexibility in routing that results because there are no alternate routes. The analysis will show that although the ring may accommodate a large throughput when high traffic points are close on the ring, there is no flexibility in adapting to redistribution of traffic requirements. The example is carried out for a mixture of tape transfers and interactive traffic.

One of the traffic models developed by Hayes and Sherman [31] is used to analyze the ring design. We consider the design of a single slotted ring to which sources of traffic are connected through an interface. The source can represent host computers, terminals, or a combination of these. The interface is assumed to receive packets from the source, store them, and multiplex them onto the ring. A header which addresses the packet to a particular interface on the ring is added to the packet; the packet size on the ring is therefore larger than that on the line. In the reverse direction, the interface removes packets from the ring addressed to it, removes the "ring header," and transmits these packets to the source. It is assumed that an interface can remove a packet from the ring and then feed a new packet into that same slot. In this case, the traffic on the ring seen by the interface is in the location marked by X in Figure 8.1. That is, it includes only the traffic

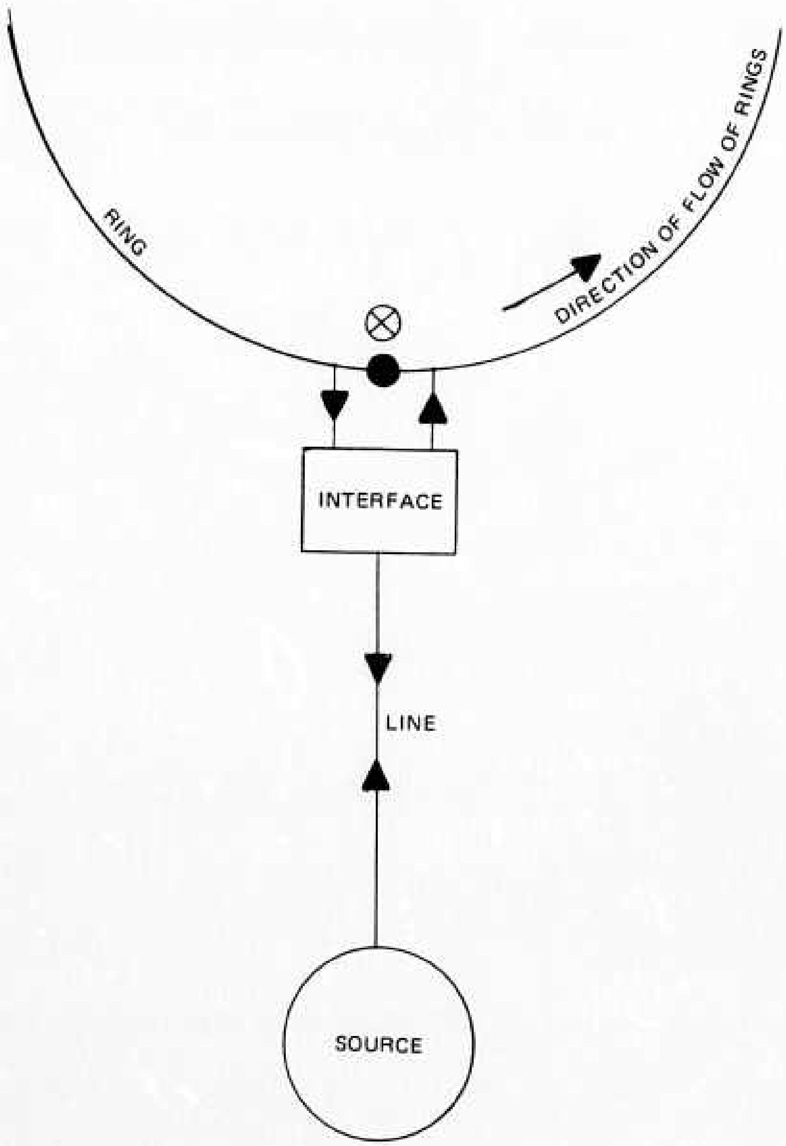


Figure 8.1: Ring-Source Interface

which passes through the interface and not the traffic which is destined for that interface or which originates from it. It is assumed that durations of idle periods on the ring are exponentially distributed.

In the calculations of the buffer content and the delay, we assume that the traffic flow from a source to its interface is at a constant rate, equal to the average rate. The performance of the system is characterized by the buffer size at the interfaces, the delays at these interfaces, and maximum throughput which can be obtained.

For all cases, we used 1.5 Mbps speed on the ring, a packet of 784 bits on the ring, and 768 bits on the lines to the interfaces. Table 8.1 gives the input data to the interfaces, and Table 8.2 gives the distribution matrix P_{ij}, that is the fraction of traffic from interface i destined to interface j. An important parameter is the average number of packets per message. We assume an average of 14 packets/message when all traffic is of an interactive type, 65105 packets/message when all of the traffic is tape transfer, and an average of (1500 ÷ 65105) packets/message for other interfaces depending on the fraction of tape transfers that it included.

For the given data we analyze two designs referred to as System 1 and System 2. System 1 connects the interfaces in order 1 through 4, and the direction of flow on the ring is counter clockwise. Interface pairs with high traffic requirements are relatively close. For System 2, the ring is reconfigured for flow in the following direction: 1,14,8,6,7,9,12,10, 4,13,11,3,2,5,1.

Tables 8.3a and 8.3b show the results for System 1 and 2. Each table shows utilization of the ring seen by an interface, the rate of packets/sec. on the ring seen by an interface,

Table 8.1: Data at Interface

Interface No.	Line Speed Bits/Sec	Rate In Bits/Sec	Rate In Rate/Sec	Average No. of Packets/Msg.	Ratio of Source To Ring Rate
1	230000.	87754.	114.3	14974.	.15655
2	100000.	59554.	77.5	65105.	.06807
3	100000.	61646.	80.3	14.	.07071
4	100000.	41554.	54.1	65105.	.06807
5	100000.	30785.	40.1	26042.	.06807
6	100000.	10154.	13.2	65105.	.06807
7	100000.	17754.	23.1	65105.	.06807
8	230000.	133754.	174.2	29948.	.15655
9	100000.	17015.	22.2	26042.	.06807
10	230000.	69754.	90.8	44922.	.15655
11	100000.	30154.	39.3	14.	.07071
12	100000.	41015.	53.4	26042.	.06807
13	100000.	25354.	33.0	14.	.07071
14	100000.	65554.	85.4	65105.	.06807

Table 8.2: Fraction of Traffic To Interface

Fraction of Traffic to NIP													
1	2	3	4	5	6	7	8	9	10	11	12	13	14
0.000	.091	.227	.066	.063	0.000	0.000	.340	.031	.055	0.000	.072	0.000	.055
.336	0.000	.358	0.000	.013	0.000	0.000	0.000	.026	.081	0.000	.026	.161	0.000
.082	.815	0.000	.004	.016	.004	.004	.004	.029	.004	.004	.029	.004	.004
.116	0.000	.056	0.000	.043	0.000	0.000	0.000	.037	.116	0.000	.037	.597	0.000
.158	.002	.077	.002	0.000	.002	.327	.002	.052	.158	.080	.052	.002	.080
0.000	0.000	.227	0.000	.076	0.000	.394	0.000	.152	0.000	0.000	.152	0.000	0.000
0.000	0.000	.130	0.000	.043	0.000	0.000	0.000	.199	.541	0.000	.087	0.000	0.000
.223	0.000	.017	0.000	.006	0.000	.486	0.000	.012	.022	0.000	.012	0.000	.223
.080	.009	.145	.009	.054	.009	.009	.009	0.000	.268	.009	.241	.009	.150
.069	.049	.067	.275	.080	0.000	.138	0.000	.036	0.000	.112	.105	.069	0.000
.318	0.000	.077	.080	.105	0.000	0.000	0.000	.051	.159	0.000	.131	0.000	.080
.121	.004	.060	.004	.023	.004	.004	.004	.041	.121	.062	0.000	.491	.062
0.000	.379	.091	.189	.030	0.000	0.000	0.000	.061	.189	0.000	.061	0.000	0.000
.073	0.000	.035	.037	.048	0.000	0.000	.381	.023	0.000	.037	.060	.305	0.000

Table 8.3: Comparison of Ring Designs

	Interface Number	Utilization Of Ring Seen By Interface	Pack/Sec On Ring Seen By Interface	Average Number of Packets In Interface Queue	Average Delay Per Packet In Seconds
Table 8.3a System 1	4	.2211	423.04	.44	.00821
	13	.2288	437.60	.28	.00854
Table 8.3b System 2	4	.1811	346.41	40.28	.74447
	13	.1550	296.59	29.95	.90726

the average number of packets waiting to be multiplexed onto the ring, and the average delay per packet. An important point to notice is that in System 2 at interface 4, the average number of packets in the queue is over 40, a severe degradation in performance caused by a redistribution in traffic requirements.

Chapter 9
CATV SYSTEMS FOR LOCAL ACCESS

9.1 Introduction

A wide variety of system configurations such as loop structures and various multiplexing schemes have been proposed for communicating data on future CATV Systems [39].

A circuit switched video system has been developed by Rediffusion International Ltd. in Great Britain [24]. Multipair cables are used with each pair being dedicated to a separate subscriber. He may then select the program of his choice by means of a telephone type dial. The Rediffusion System presents interesting tradeoffs between initial investment, flexibility and reliability. However, since this type of system has not made significant inroads into the U.S. market at present we will not consider it further here.

We first present a very brief introduction to the structure common to most of the 3000 current U.S. CATV Systems [57].

Signals are received at an antenna located for ideal reception and are then relayed from this "head end" to individual subscribers via a distribution system of coaxial cables, broadband repeater amplifiers, and subscriber taps.

A cable television distribution system generally consists of a trunk section and a feeder section. The trunk section contains trunk cable connecting the head end to distribution points, from which the feeder cable emanates. Located along the trunk cable are high-quality repeater amplifiers, which provide gain along the trunk and to the feeders. At the termination of the trunks there are distribution amplifiers. Along the feeder cable there are lower quality amplifiers called extender amplifiers and subscriber taps that provide signals to drop cables leading to home receivers.

With recent broadband amplifiers, the full Sub-UHF spectrum from 5 to 300 MHZ has been used. Partitioned into 6 MHZ channels for television, only a small amount of this spectrum is currently used for TV signals.

FCC regulations now require that new CATV Systems must have two-way capability. Practically speaking, this does not mean that all new systems are two-way systems, but rather that amplifier units are installed with forward amplifier modules in place and with distances between amplifiers constrained so that at some future date reverse amplifier modules can be installed for two-way operation. However, a number of actual fully two-way systems are presently being built and the number is increasing rapidly. Most present two-way systems use the configuration in Figure 9.1a. Filters at each end of the station separate low (L) and high (H) frequencies and direct them to amplifiers. Two possible "two-way" configurations [33] are shown in Figures 9.1b and c.

Of course, two-way CATV Systems are themselves in an experimental stage so that there are still implementation problems in achieving written specifications. Some of the classical electrical and communication bugs are being removed at present—ringing around loops through band separation filters, tuning return AGC's, alignment procedures and construction problems.

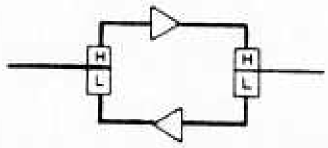


Figure 9.1a: Two-Way CATV Repeater

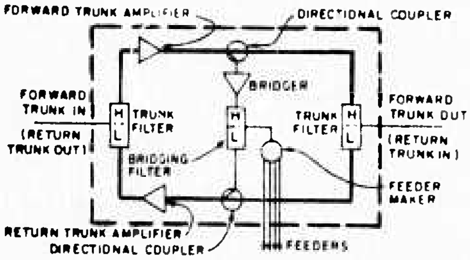


Figure 9.1b: Two-Way CATV Repeater (With Feeders)

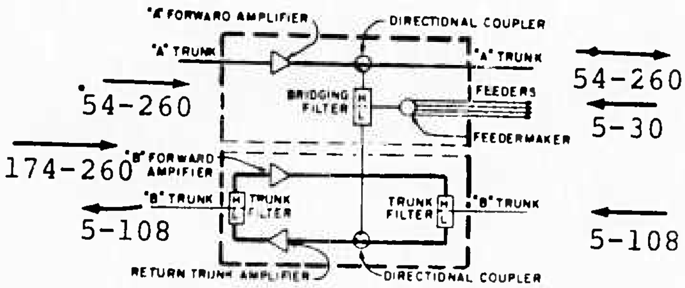


Figure 9.1c: Dual Trunk/Single Feeder Station
(Suburban Boston Configuration)

A number of companies have developed system concepts and subscriber hardware to implement digital home response modes for existing CATV Systems [7, 8]. Among these are Theta-Com, Jerrold, Rediffusion Electronics, CAS Manufacturing Co., Hughes Aircraft, AMECO, Scientific Atlanta, and Cable Information Systems. Several of these companies are running prototype systems in cities throughout the U.S.—El Segundo, California; Dennisport, Mass.; and Orlando, Florida, among them. In addition MITRE of McLean, Va., has installed an experimental system in Reston, Va., which incorporates a "frame grabbing facility" to enable the viewer to store a frame of video data produced by a character generator. Data frames are sent every 1/60th of a second interlaced with standard video frames [56].

In most of these systems an FSK or PSK signal occupying a 4 MHZ bandwidth is used at about a 1 megabit per second rate with different carrier frequencies to and from the central antenna site. In each case, customers are polled at regular intervals to determine access to the channel. Typical proposed uses of these systems are opinion polling, meter reading, shopping, systems diagnostics and alarms. Acceptable response times are in the order of several seconds or in some cases, even minutes [30]. We will investigate data transmission on existing CATV systems with required response times of tenths of seconds and with up to 100,000 interactive users.

To illustrate our points in detail, we will consider a specific design for the Suburban Boston complex. The design will use the "feeder backer" configuration shown in Figure 9.1.c with the frequencies assigned as specificly indicated. The design techniques for the CATV Systems themselves are well known applications of classical communications techniques [16, 21].

9.2 Data Error Rates on CATV Systems

Two-way CATV systems permit input from virtually any location in the network. The result is a large number of noise sources being fed upstream toward a common source. CATV amplifiers have a noise figure of about 10db for a 5 MHz channel. Cascading amplifiers can increase effective system noise figure by 30db or more. Nevertheless, we shall see that system specifications on signal-to-noise ratio for CATV systems are stringent enough so that data can be sent with existing analog repeaters, and no digital repeaters, such that bit rate error probabilities are negligible.

For example, if the worst signal-to-thermal noise ratio is limited to 43db and the worst cross-modulation to signal ratio is limited to −47db, system operators may want to limit data channel carriers to a level of 10 to 20db below TV operating levels in order to minimize additional loading due to the data channel carriers [51]. Accepting these restrictions, in the worst case, we would be limited to 23db signal to thermal noise ratio and −27db cross-modulation to signal ratio. Let us consider both of these sets of restrictions to determine the resulting CATV system performance for random access packet transmission.

We calculate error rates for a FSK system with incoherent detection to determine a lower bound for system performance. The error rates for coherent detection or phase shift keying, of course, would be even lower.

Let S = Signal power
N = Noise power
N_c = Cross-modulation noise power
N_r = Thermal noise power
t = Average synchronization error time
T = Bit width time

Then the sign to noise ratio is:

$$\frac{S}{N} = \frac{Sq}{N_c + N_r} = \frac{q}{(N_{c/s}) + (N_r/S)}$$

where $q = (1 - \frac{2t}{T})^2$

Let P_e be the bit rate error probability.
Let m be the number of keying frequencies in a multiple FSK system.
Then [49]:

$$P_e = \frac{m-1}{m} e^{\frac{-(S/N)}{2(m-1)^2}}$$

We assume that each packet carries its own synchronizing bits and hence there is no need to synchronize every terminal to a master clock. Therefore, temperature, pressure, and humidity variations which have approximately the same effects at all frequencies do not enter into the calculation of t. The group delay variation over a six Megahertz bandwidth is less than .2 μseconds [48]. For a 1 Megabit pulse rate T = 1 μsecond and t = .2 μseconds. Hence, q = .36. We, therefore, have the error probabilities in Table 9.1 for the suburban Boston complex.

For effective signal-to-noise ratios above 20db there is a threshold effect for error probabilities. This is borne out by the negligible error rates. Even for the degraded specifications the error rate is low enough for the most stringent practical data requirements. Furthermore, at a rate of 10^6 pulses/second the FSK signal will occupy the 6MHz bandwidth with negligible intermodulation into TV channels.

Consideration of reflections, intersymbol interference and 60 cycle hum also lead to the conclusion that CATV systems are excellent media for packet data transmission.

The signal levels in a CATV system are controlled via AGC and dual pilot carriers. Ripples are kept to less than 1db over the whole frequency band. In any case, frequency shift

Table 9.1: Error Rates for FSK

System Label	Type of Specification	(N_c/S)	(S/N_r)	m	P_e
A	Undegraded Specs.	-47db	43db	2	$1/2e^{-5,148} \approx 1/2 \times 10^{-2,239}$
B	Undegraded Specs.	-47db	43db	4	$3/4e^{-571} \approx 3/4 \times 10^{-248}$
C	Undegraded Specs.	-47db	43db	8	$7/8e^{-104} \approx 7/8 \times 10^{-45}$
D	Boston Specs. - degraded by 20db	-27db	23db	2	$1/2e^{-51} \approx .5 \times 10^{-22}$

keying is insensitive to small amplitude variations. The effect of group delay error has already been taken into account in the use of q in the formula for error probability. The remaining source of intersymbol interference is the reflection of pulses and the effect of the reflected pulses on the transmitted data. There are three types of disturbances due to reflections. In each case, we shall see that video restrictions are certainly stringent enough to avoid any difficulties for data transmission.

Periodic changes of minute magnitude uniformly distributed along the cable length, the magnitude of changes being essentially equal from period to period, due to the nature of the manufacturing process, cause reflections which add in phase at certain frequencies. The signal strength relationship of the reflected wave to the incident wave is referred to as structural return loss (SRL). Typical values for the magnitude of SRL are better than -26db [43].

Assuming that the reflected signal is always of an opposite sign to the original signal, the signal level is degraded by at most S–a, where a is the amplitude of the reflected signal. The signal-to-noise ratio becomes [53]:

$$\frac{S-a}{N} = \frac{S}{N}\left(1 - \frac{a}{S}\right)$$

In other words $\frac{S}{N}$ is degraded by $\left(1 - \frac{a}{S}\right)$

For a reflected signal of -26db, $\left(1 - \frac{a}{S}\right)$ is .9975—quite acceptable.

A localized change or changes on the cable cause echo phenomena. Low reflection coefficients of active and passive devices and the use of directional couplers at all subscriber taps ensure that the magnitudes of reflected pulses are in the "no ghost range" of Figure 9.2 [45, 40]. These are translated into critical distances for different types of cable in Figure 9.3. Thus, for example, considering the reflection on .412 inch cable at Channel 13 the critical distance is about 250 feet and the ratio of the magnitude of the reflected signal to the magnitude of the original signal is -23db.

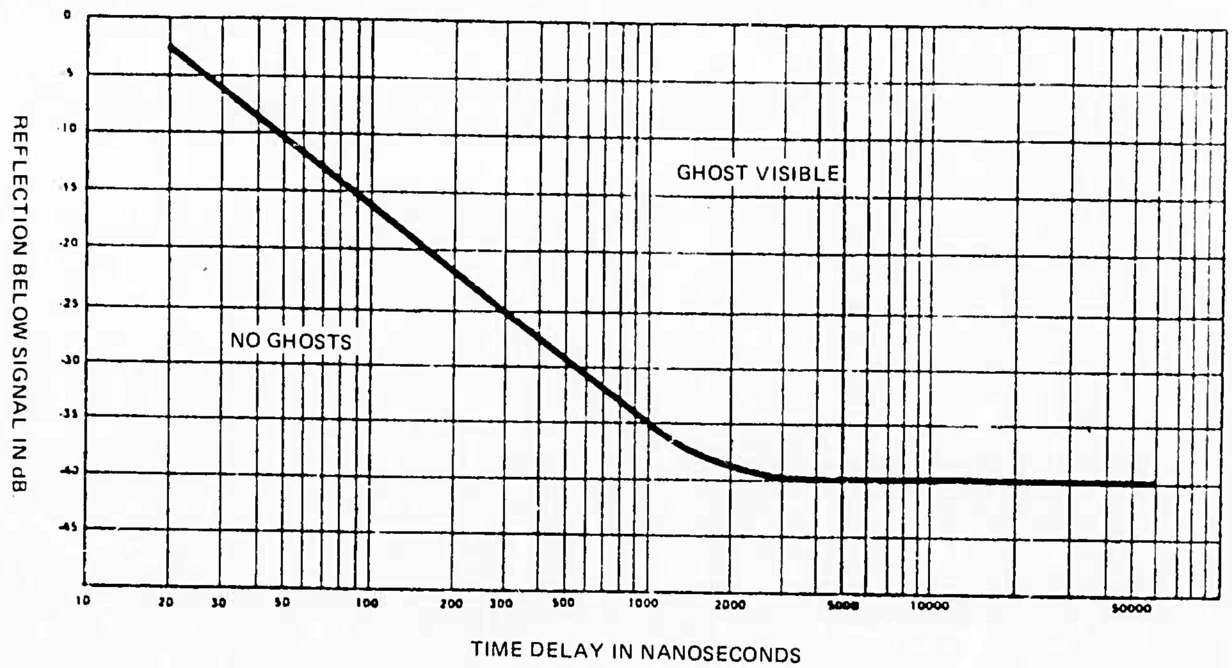


Figure 9.2: Curve Showing Perceptibility of Ghosts

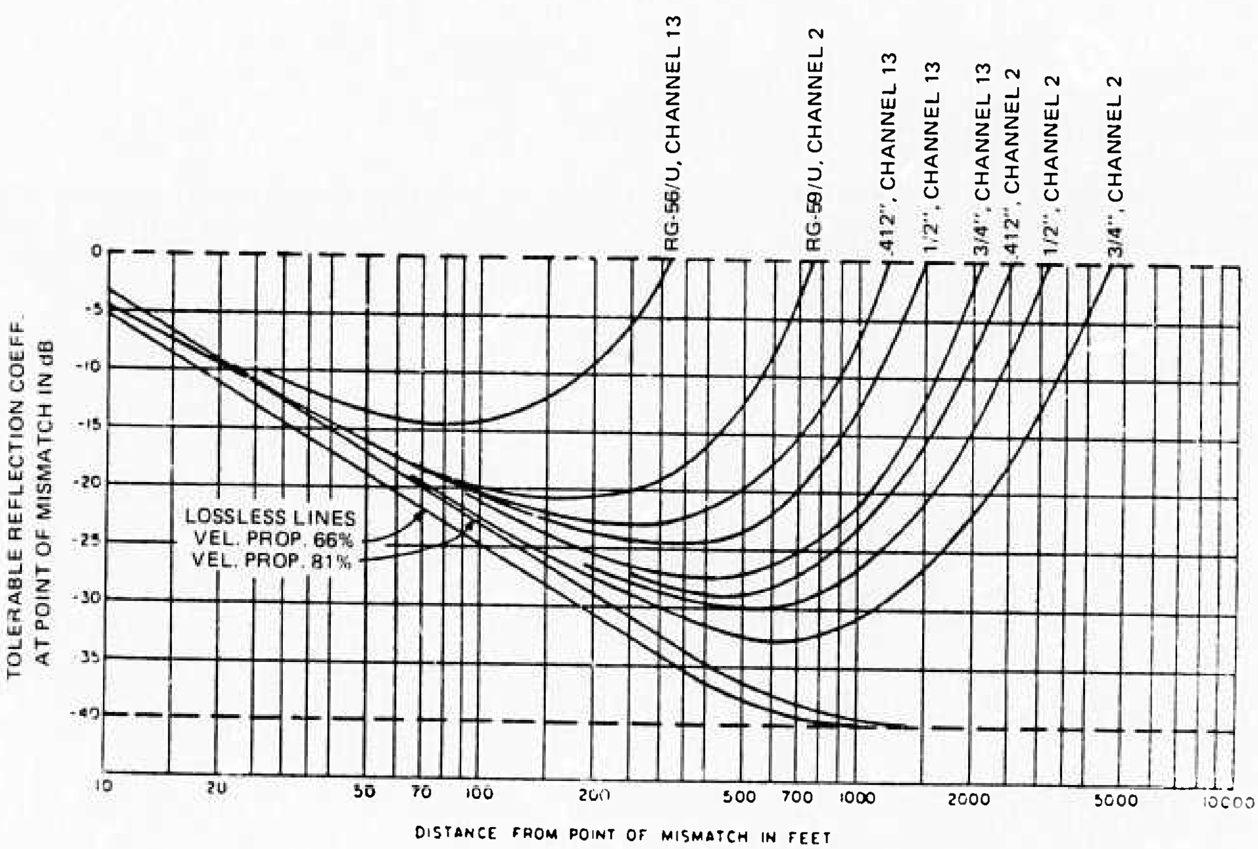


Figure 9.3: Graph for Determination of Critical Cable Lengths

Randomly distributed changes of random magnitude which persist throughout the cable length cause reflections which do not add in phase. These can be taken into account in noise calculations and are usually negligible.

Cable system amplifiers are powered by low voltage 60 Hz power through the co-axial cable. This power may be as high as 60 volts (RMS) and currents may run to 10 ampheres (RMS) with peak currents even higher. There are significant harmonics of the power line frequencies present. Some amplifiers use switching mode power supplies with switching frequencies in the 10-20 KHz range. Hash from these switching regulators also finds its way into the cable. However, both the 60 cycle harmonics and hash limit only the area of very low frequencies which are generally avoided for data transmission.

9.3 Other Performance Criteria for CATV Systems

Data users may find cable system reliability quite poor when compared with the common carrier facilities with which they are familiar. One of the major problems with data transmission on CATV Systems is that there is no redundancy of path cables or amplifiers. There are no government or industry minimal standards for acceptable performance; hence, performance will vary from system to system. Many old systems were built to extremely loose specifications on noise and cross-modulation and have serious reflection problems because of the use of unmatched subscriber taps. Fortunately, systems in large cities and new buildings are much newer and are required to meet more exacting standards.

Even with these systems, the construction norms are still those which satisfy casual TV viewers, not data users. Thus, loose connections cause intermittant transmission conditions, and momentary "disconnects." These would cause only minor "flashes" on a TV picture but constitute major data dropouts in a high speed data circuit. Finally, systems may be inadequately tested, and hence, in some parts of a CATV System noise and cross-modulation levels may not meet written system specifications. The limiting factor in determining the performance of the system will not be Gaussian noise interference, but a number of practical factors which provide interference, generally categorized as "impluse noise." These factors are difficult to characterize and include phenomena such as loose connections, cracked cable sheaths, and R-F leaks.

Two factors dominate the specification of any data transmission mode on a CATV System.

a. That data is sharing a transmission medium with video signals.

b. That there will be a large number of users.

9.3.1 Interface with CATV System

Two Way Options

The data transmission system must be readily adaptable to a wide variety of existing CATV System designs and two-way options.

Data Rates

The data signals must not cause visible interference with video signals.

Installation

If auxiliary data equipment is to be added to the CATV System, it must satisfy the following requirements:

- It can be installed with only minor changes in the CATV System.
- It need be installed in only a small number of locations.
- It can be installed rapidly in early hours of the morning to prevent interference with TV service.

Low Cost

To maximize the marginal utility of data distribution over the CATV System, any equipment introduced must be inexpensive.

9.3.2 Interface With Population

Population Density Variations

Standard transmission configuration options must be available for systems of various sizes, population densities and percent of active users. Because of the huge number of potential users, all terminal equipment must be simple and inexpensive.

Unsophisticated Users

To minimize user interaction with the system operating mode, all terminal equipment must be the same for each location; it must use the same frequencies and data rates; and it must have no options for equipment modification by the user.

The MITRE Corporation has patented a system called MITRIX which meets all the above specifications [58] and has many other excellent features. Some of the disadvantages of polling for terminal-oriented networks [50] are the synchronization delays [32] and the large amount of channel bandwidth occupied by simply polling 100,000 subscribers. MITRIX overcomes both these problems by using a time division multiple access scheme. Furthermore, since the number of time slots per second is dynamically assigned to a subscriber, the system also avoids the wasted bandwidth in allocating fixed frequency bands (FDM) or fixed time-slots (TDM) to subscribers who are active for only small amounts of time. The users interface unit requests a certain number of time slots within frame periods and these are then allocated by a Computer Digital Interface Unit; a DEC PDP-15. The system is highly flexible in structure, efficient and inexpensive.

9.3.3 Other Considerations

However, there are still some tradeoffs involved and for large systems improvements are still in the offing. In particular, we are still faced with the problem that if a subscriber logs in at a terminal and makes a request for a certain data rate then he holds those time slots until he logs out whether he is actively typing or thinking and not typing. For small systems or systems with a small number of users, this may be an acceptable inefficiency in bandwidth use. But for systems like the suburban Boston system it may not be acceptable. As we shall see the factors involved are the available bandwidth, the average ratio of active user time to inactive user time in a logged-in period, and the average number of active users. The alternative which makes more efficient use of bandwidth for high peak to average data rates is the random access packed multiplexing method previously derived and applied to the satellite channel. With packets, terminals seize the channel only when they are active. Hence, more users can be accommodated. Furthermore, reservations of channels are not required. As we have seen the random access feature results in a channel availability of 1/2e of the band or 1/e for a slotted system. This is effective if the average peak to average data rate is greater than the number 2e or e, respectively.

9.4 Number of Active Terminals for Random Access Packet Sample System

We have seen in Chapter 5 that for an unslotted random access packet channel the maximum number of active terminals k_{max} is given by $(2e\lambda\tau)^{-1}$. If we let d be the pulse duration and let γ be the number of pulses per packet, then k_{max} is $(2e\lambda\gamma d)^{-1}$ where $\lambda\gamma$ has the dimension of pulses/second per terminal. For a two level FSK system, this is the same as bits/second per terminal. In Figure 9.4 we plot the maximum number of active terminals versus $\lambda\gamma$ for the systems in Table 9.2 using the above equations. The curves labeled A, B,C, and D correspond to the slotted system in Table 9.2 labeled A,B,C, and D. The lines labeled A', B', C', and D', are for the corresponding unslotted systems.

We can now examine Figure 9.4 to determine system performance under some typical data transmission requirements. For a data rate of 40 bits/second per terminal, a single trunk can handle 4000 terminals with a slotted system (1 Megabit/sec) with an error rate of 10^{-22} at a signal to noise ratio degraded by 20db. The average number of TV sets per trunk in the suburban Boston system is approximately 27,000. Hence, the simplest

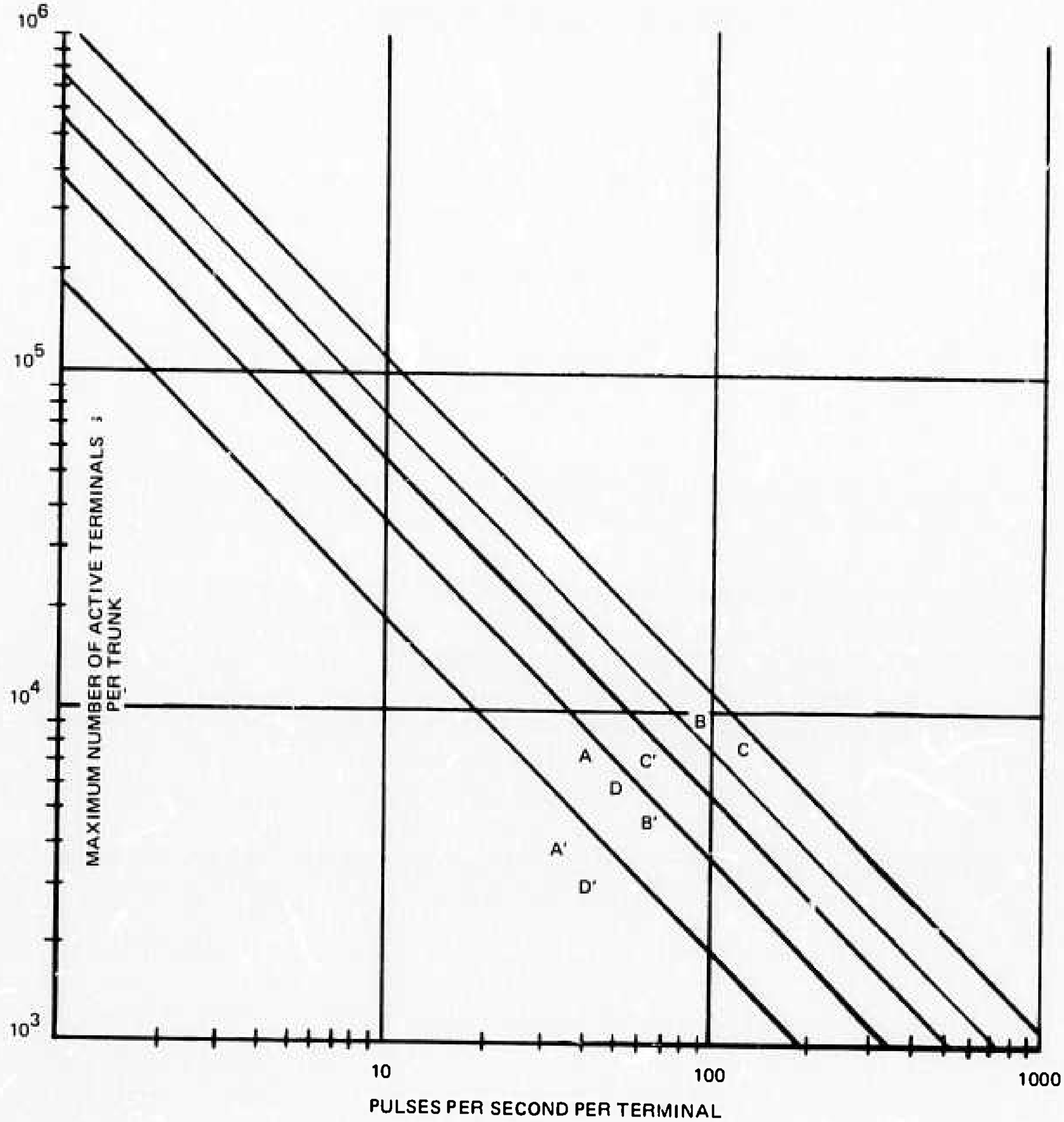


Figure 9.4: Performance of Random Access Packet Cable System

modulation scheme will handle one third of all terminals as active terminals. At 100 Kbits/second, the system will handle 900 active terminals.

The reason for considering a 100 Kbits/second channel instead of 1 Megabit/second channel is that it allows more adaptability for local point-to-point traffic with the addition of standard digital devices such as concentrators or routers. Of course, if this is not required, the 1 Megabit channels can be used.

Table 9.2: Number of Active Users Per Trunk

Type of System / Data Rate	Slotted System	Unslotted System
1 Megabit/sec.	9,000	4,500
100 Kilobit/sec.	900	450

We will use the terminology of the cable TV industry in describing the direction of signal flow. Signals traveling from the head end toward terminals will be said to be directed in the "forward" direction on a "forward" link and signals traveling from terminals toward the head end will be said to be directed in a "reverse" direction on a "reverse" link. A convenient synonym for "forward" will be "downstream," and for "reverse" wil be "upstream." To install the device to be described in the forward and reverse channels simple duplex and triplex filters can be used. The devices are illustrated schematically in Figure 9.5.

9.4.1 Carrier Frequency Conversion

In the simplest version of a data system, two carrier frequencies are used; one for forward transmission from the head end to the terminals, and one for reverse transmission from the terminals to head end. Let us call these angular frequencies ω_f and ω_r respectively. The next simplest option is to use frequency converters at a small selected set of points in the system. In the forward direction, the converter converts from ω'_f to ω_f and in the reverse direction, it converts from ω_r to ω'_r. The net result is that the terminals still receive and transmit at the frequencies ω_f and ω_r. However, in the trunk between the converters and the head end, there are four frequencies in use, ω_r, ω'_r, ω_f and ω'_f so that in these trunks twice the traffic can be handled.

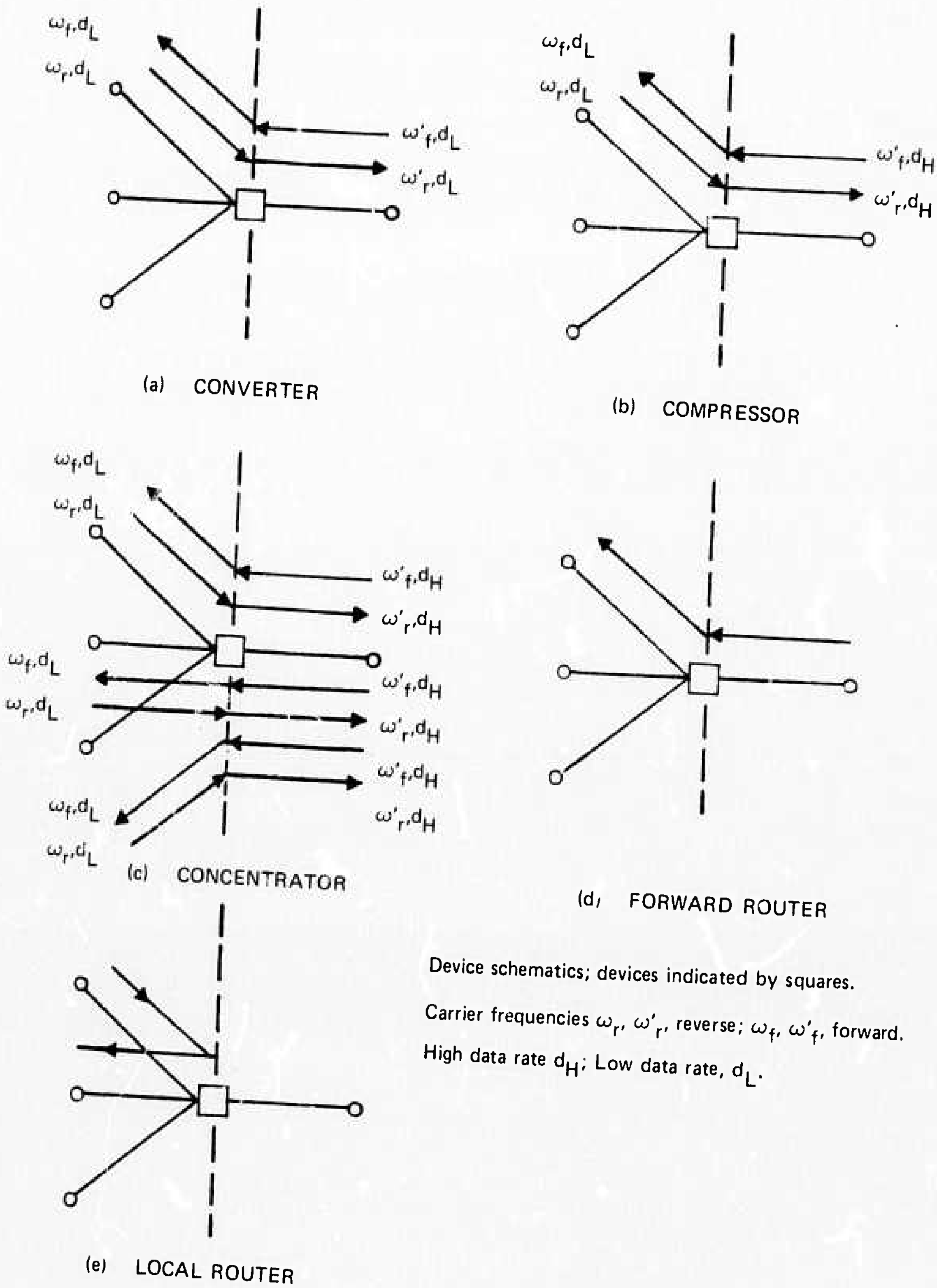
The advantages of this scheme are:

a. All terminals are identical.

b. The capacity of the system is increased since two channels are available in each direction for heavy traffic sections of the cable.

9.4.2 Routing

There is no requirement for routing since a basic premise is that all receivers listen to all messages that reach them and merely select the ones addressed to them. Nevertheless, we will consider the addition of some primitive low cost routing schemes and qualitatively indicate their effect on system capacity.



Device schematics; devices indicated by squares.

Carrier frequencies ω_r, ω'_r, reverse; ω_f, ω'_f, forward.

High data rate d_H; Low data rate, d_L.

Figure 9.5: Concentration Alternatives for Packet Cable System

In the central transmission mode there is no routing needed in the reverse direction since all messages reach the head end along the unique paths from the originating terminal. In the forward direction, the signals at frequency ω_f are blocked by filters at the converters and yields a simple form of routing. In fact, at any section of trunk not requiring signals at ω'_f filters can be added to block ω'_f. Adding these filters does not increase system capacity but may be useful if the frequency ω'_f can be used for local signaling when not being used for data transmission.

At any junction containing a converter, digital routers *must* be added to send messages at ω_f down the trunk to which they are addressed rather than all trunks. Such a router may be added at any other point in the system as well. This is called *"forward routing"* and can increase system capacity. Forward routing requires a digital router which can read and interpret message addresses.

Let us now consider these system options in the presence of local traffic. The option of frequency conversion is unaffected and performs in exactly the same manner as in the central transmission mode. However, an extra routing option is available for local transmission. In particular, if two terminals are on the same trunk, then the message between them can be intercepted and routed at a routing station rather than travel all the way to the head end. Such routing is called *"local routing."* Local routing reduces the traffic on the main trunk.

9.4.3 Compression

As the next more complicated option, the data rate as well as the carrier frequency is changed at a converter, i.e., a compressor can be used. The advantages of this arrangement are:

a. All terminals operate at low data rate.

b. On heavily used lines near the head end, a higher data rate, say one megabit/second, can be used to increase the number of potential active users or decrease the delay.

c. Even though a section of the trunk can carry high data rate traffic at carrier frequencies ω'_r and ω'_f, other users can still use the system at the low data rates at ω_r and ω_f. Thus, the number of compressors required is small.

9.4.4 Concentration

Finally, the compressor at junctions may be replaced by a concentrator. That is, messages arriving simultaneously on two or more links in the reverse direction are buffered and sent out sequentially at the higher data rate. This essentially makes the system downstream from the concentrator appear to operate at the higher data rate and hence increases the system capacity even further.

9.4.5 Frequency Division Multiplexing

In case the data rate is limited by the head end mini-computer, an available option is to frequency division multiplex several 100 Kbits/second channels, each of which is processed by a separate head end minicomputer.

The assignment of these options in an optimal fashion requires detailed expressions for the traffic in the links. Formulae for the traffic in links using combinations of the various digital devices are fairly obvious in detail although rather tedious to present in generality.

9.5 Random Access Packet Designs For Sample CATV System

We will illustrate the usefulness of the various options and devices we have considered, for example, in adapting to a low data rate channel system of 100 Kbps. We apply the devices to a design of a random access packet data system for Medford, Massachusetts, a section of the Suburban Boston CATV complex. In Figure 9.6, a branch of the trunk is drawn for Medford. The triangles represent bridger amplifiers. These amplifiers feed into feeder cable and extender amplifiers with customer taps and drop lines emanating from the feeder cable. The feeder backer arrangement previously mentioned is used. The feeder system emanating from a given bridger amplifier is called a cluster. The number next to each amplifier gives the number of terminals in the cluster associated with that amplifier. The average number of terminals per cluster is 137 with complete coverage of all homes.

We focus our attention on one trunk in the Medford area. We assume that the average traffic per terminal is 40 bits/sec. We assume most conservatively that in the design the data from the terminals is at 100 Kbps; the upconverted rate is 1 Megabit/sec. We have already obtained the results in Table 9.2 showing the number of active terminals that can be supported on a trunk at each data rate.

Since we have not presented relative costs of converters, concentrators, multiplexers and routers, we are not optimizing the design. We are merely presenting feasible designs to demonstrate the wide range and flexibility achieved by combinations of a few devices. The designs are easily described by simply indicating the location of the various devices on the map in terms of an alphabetic label on the map. To aid in visualizing the design, the number in the rectangle beside the letter (on the key maps) indicates the population downstream from that point. The designs are as shown in Table 9.3.

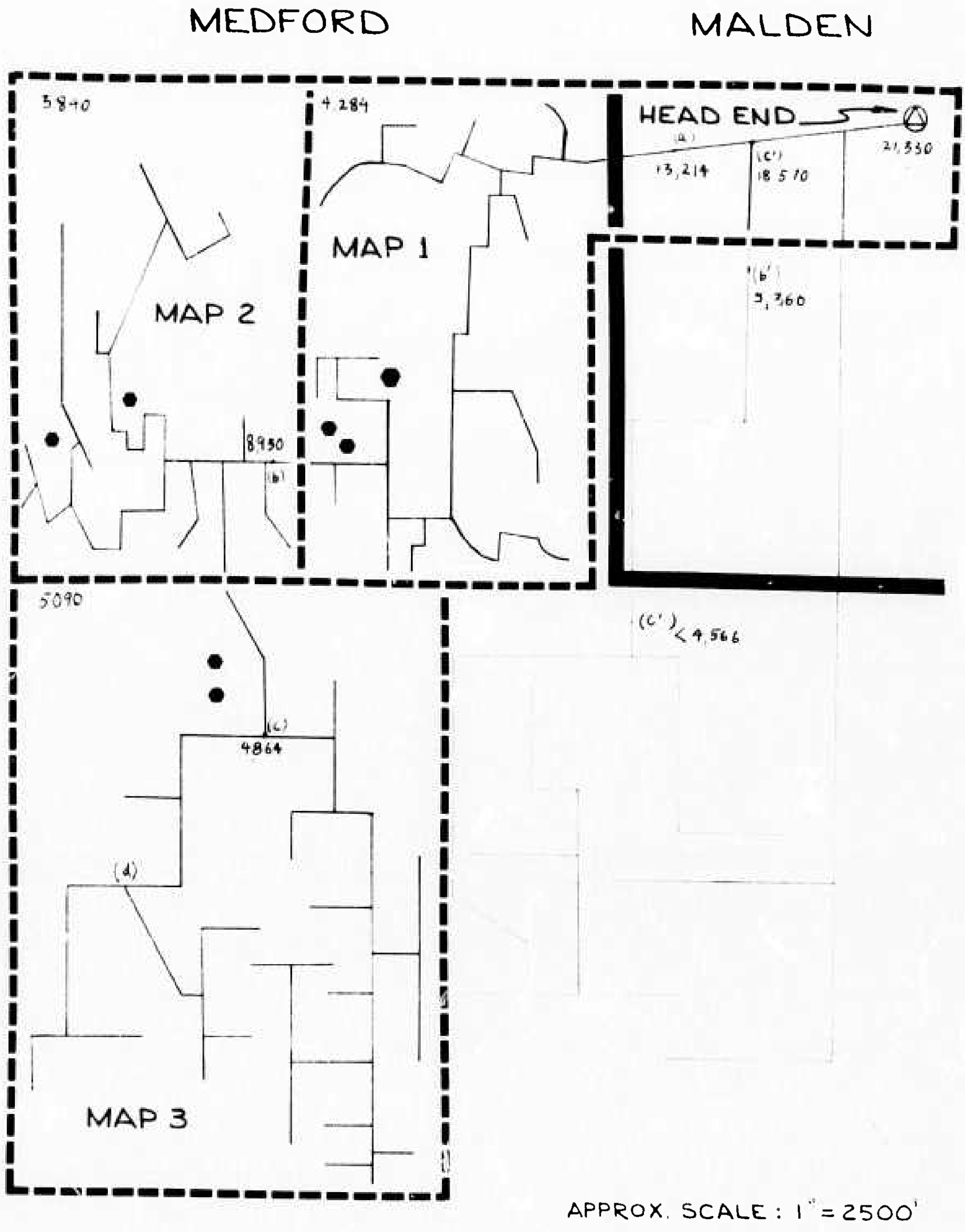


Figure 9.6a: Key Map

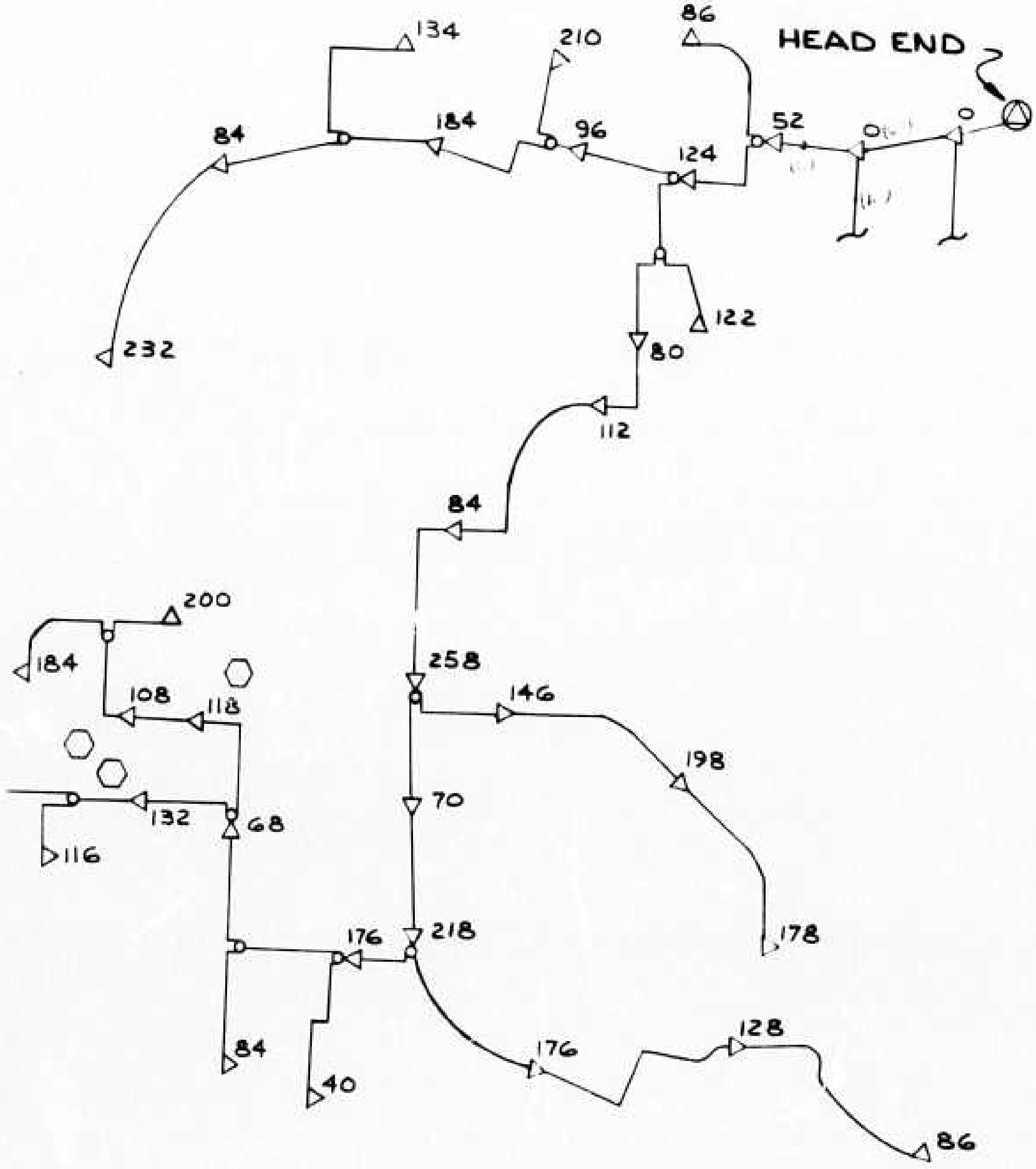


Figure 9.6b: Map 1

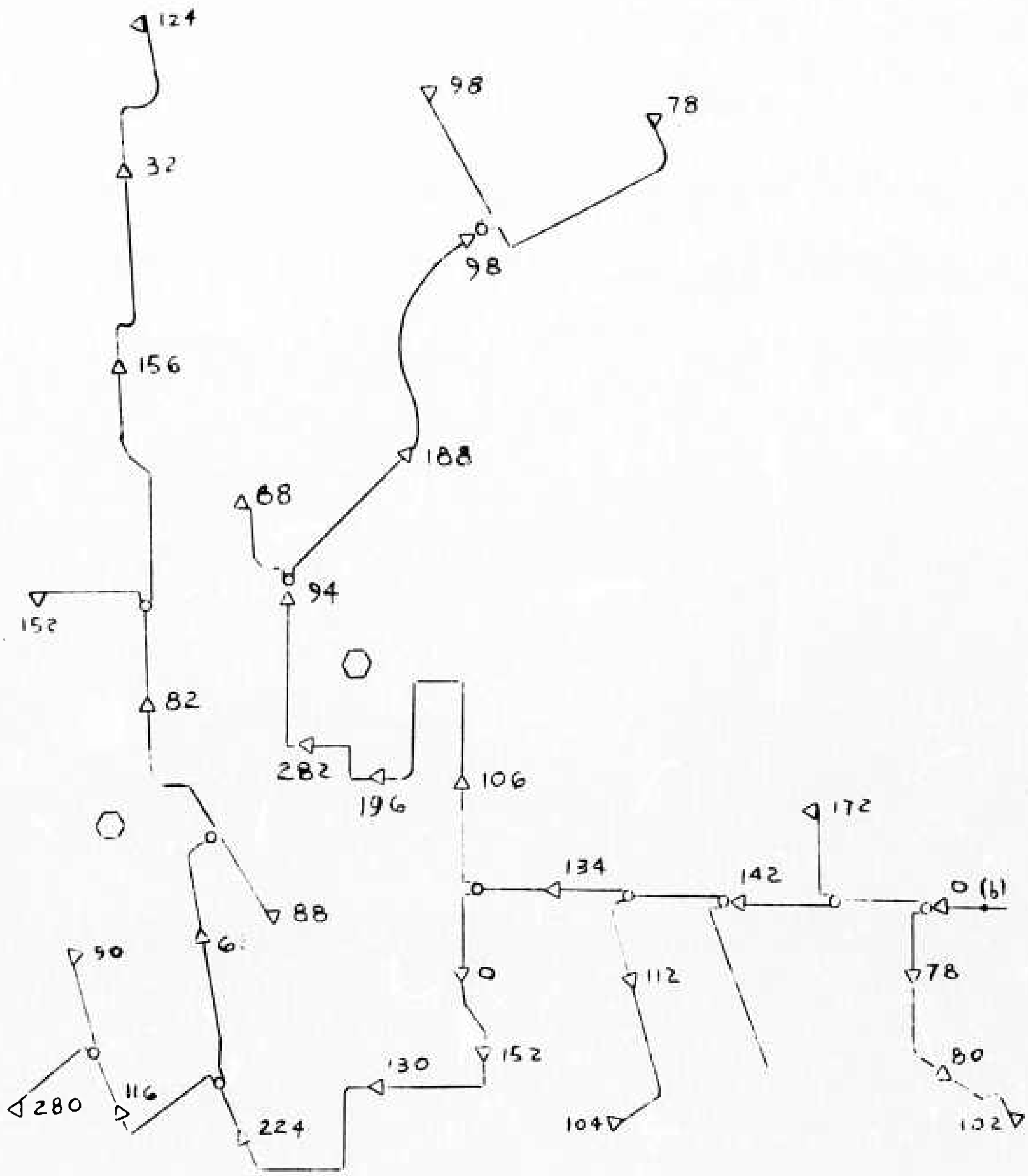


Figure 9.6c: Map 2

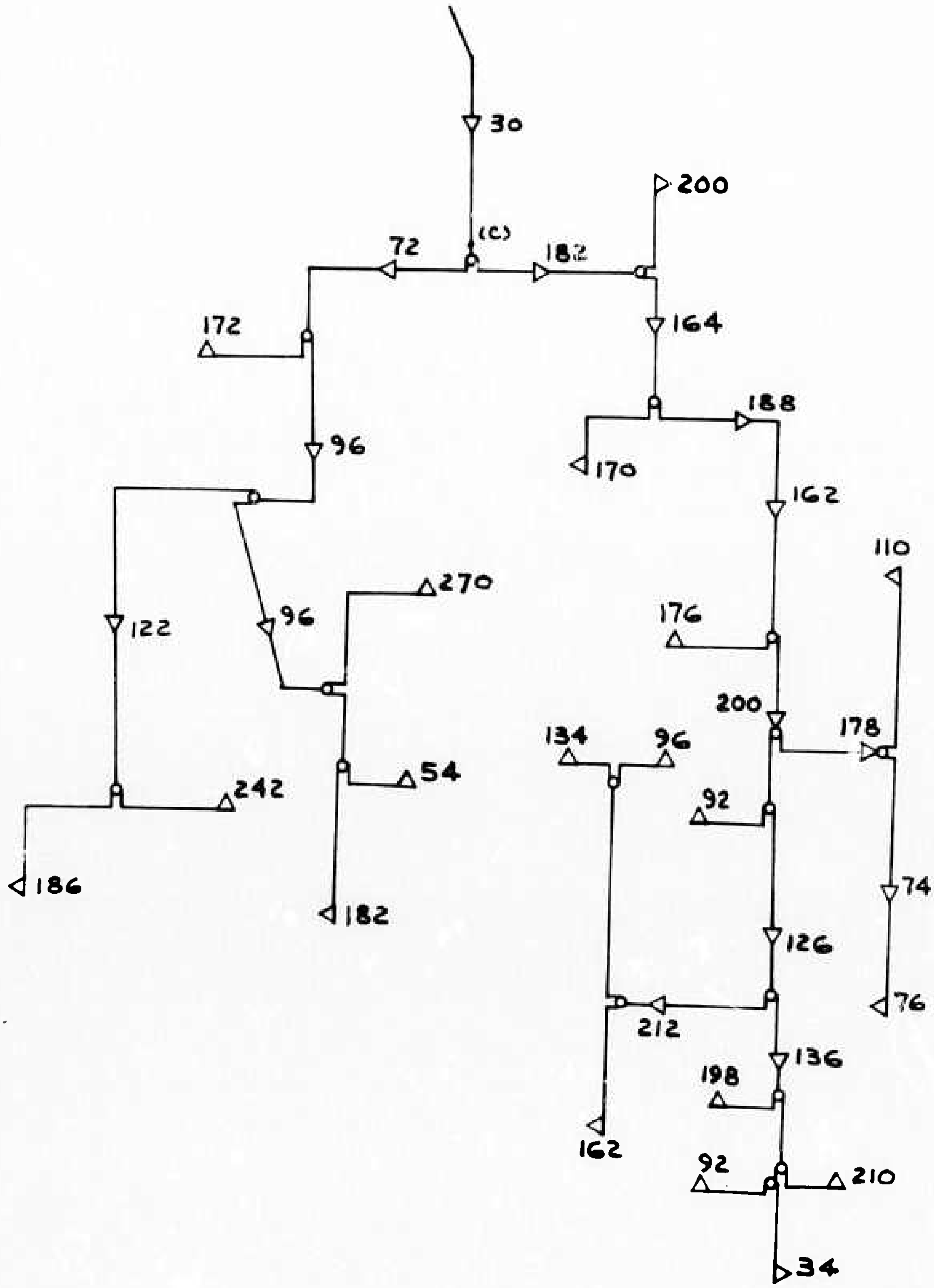


Figure 9.6d: Map 3

Table 9.3 Feasible Designs for Central Transmission Mode At 100 Kbps Data Rate

% Active Terminals	**Slotted**	**Unslotted**
1%	No devices	No devices
3%	No devices	Converter at (a)
10%	Compressors at (b) & (b′)	Compressors at (c′′) and Concentrators at (b), (c), and (c′)
15%	Compressors at (c′′) and Concentrators at (b), (c) and (c′)	

Chapter 10
PACKET RADIO NETWORKS FOR LOCAL ACCESS–INTRODUCTION

10.1 Network Overview

The main features which distinguish the Packet Radio System from a point-to-point packet switching system (such as the ARPANET) are: (i) devices in the system transmit packets by using a random access scheme, and (ii) devices broadcast so that packets can be transmitted to several devices simultaneously, and/or several packets can be simultaneously received by a receiver due to independent transmissions of several devices. These features have a major impact on practically every aspect of network considerations.

There are three basic functional components of the Packet Radio System: the Packet Radio Terminal, the Packet Radio Station, and the Packet Radio Repeater. (See Figure 10.1.) Packet Radio Terminals will be of various types, including personal digital terminals, TTY-like devices, unattended sensors, small computers, display printers, and position location devices.

In some applications the Packet Radio Station will be the interface component between the broadcast system and a point-to-point network. As such it will have broadcast channels into the Packet Radio System and Link channels into the point-to-point network. In addition, it will perform accounting, buffering, directory, and routing functions for the overall system.

The basic function of the Packet Radio Repeater is to extend the effective range of the terminals and the stations, especially in remote areas of low traffic, and thereby increase the average ratio of terminals to stations. A more detailed discussion of the network hardware functions can be found in Section 10.2.

The devices (repeaters, stations, and terminals) of the Packet Radio System communicate in a broadcast mode using a varient of the Aloha random access method [1].

Stations will be allocated on the basis of traffic. Thus, to first approximation, we can think of partitioning the area to be covered into regions of equal traffic and allocate one station for each region. In regions of low traffic density, the station may not be in "line of sight" of all the terminals in the region; hence repeaters are used to relay the traffic to the station. Thus, repeaters correspond to a geographical partition of the area into sections

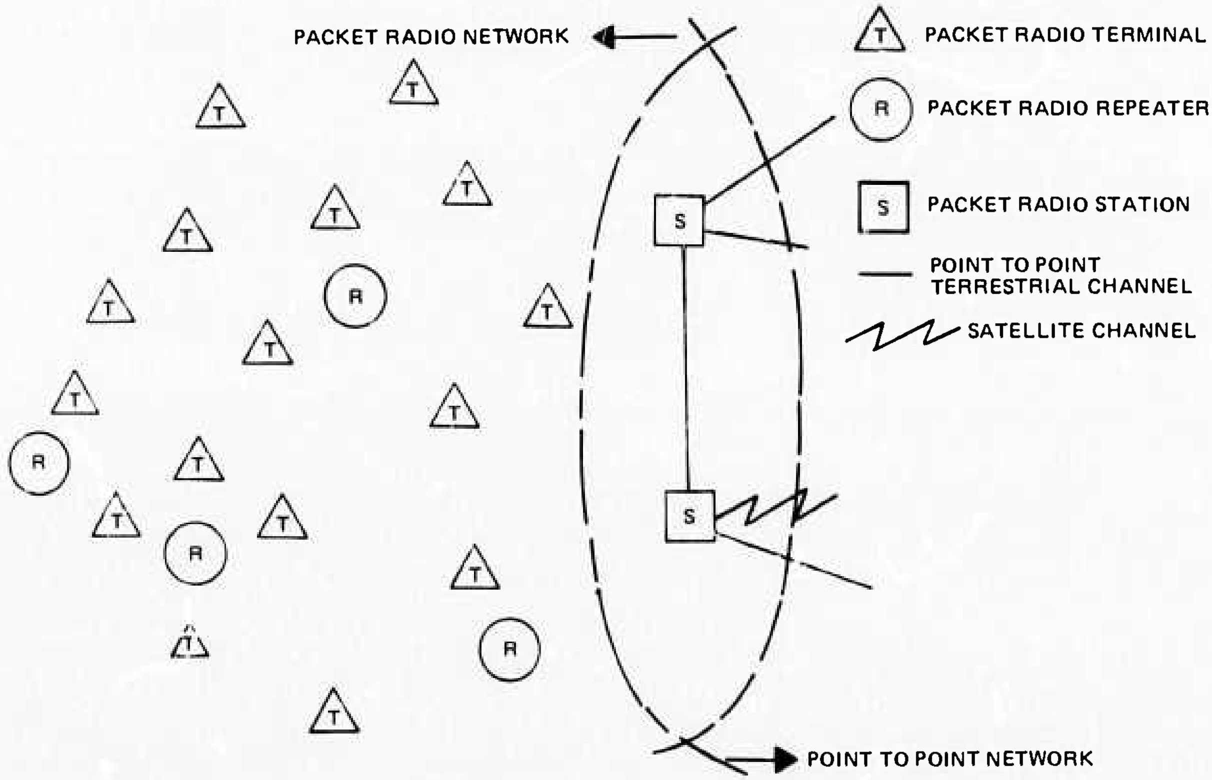


Figure 10.1: Packet Radio System

small enough so that each terminal can communicate with a repeater and its messages be relayed by repeaters to a station.

In areas of high traffic, such as urban areas, repeaters may not be needed: in fact, the problem may be that a station can communicate with more terminals than it can handle. Broadcast of data in urban areas is also complicated by multipath interference [52]. The rapidly expanding Cable Television Systems within urban areas offer an attractive alternative to over-the-air broadcasting, except for mobile users who must use broadcast techniques. As we have seen, the same general packet radio concepts can be applied to broadband Cable Systems.

10.2 Network Elements

10.2.1 Nodes

In this section we discuss the devices' functional capabilities which are necessary for *communication* in the Packet Radio network. Functional requirements of these elements not directly related to communication are not discussed.

Terminals

There are two categories of terminals; (a) those which usually await a response to a message they transmit (e.g., manually held radio terminals, small computers), and (b) those which do not require such responses or acknowledgements (e.g., unattended sensors, position indicators). Some terminals in the former category will usually send and/or receive several packets in one message.

Necessary or desirable communication capabilities of a terminal:

a. Ability to identify whether the packet is addressed to its ID.

b. Calculation of packet checksum.

c. Capabilities related to packet routing such as; retransmitting packets when acknowledgements are not received, recording and using a specific ID of a repeater and/or station to be used for other packets of the same message, counting the number of retransmissions.

d. Capabilities related to the response to previously determined types of error.

e. For unattended terminals, capabilities by which a centralized control or a station will be able to identify whether the terminal is operative or dead.

Repeaters

Functional capabilities for repeaters include:

a. Calculating packet checksum.

b. Packet storage and retransmission.

c. Capabilities by which a station can determine whether a particular repeater (or any repeater in a particular area) is operative or dead.

d. Capabilities 1, 3, and 4 of terminals.

e. Capabilities, dependent on the routing strategy, for calculating the most efficient next repeater on a transmission path to a station or to a terminal.

Stations

Among the stations' functional capabilities are:

a. A directory of terminals and repeaters in its region.

b. Operations necessary to convert packets from the Packet Radio System into packets used in the point-to-point network and conversely.

c. Storage buffers for packets received from terminals and for packets to be transmitted to terminals.

d. Storage for character position information for active terminals which do not have this capability.

e. Accounting capabilities.

f. Capabilities related to routing, flow control, and network management.

10.2.2 Channels

Communication between devices is by broadcast, using a variant of the ALOHA random access method. Many aspects of the broadcast channel are of peripheral interest in the network design of the system; however, some factors are crucial for determining the behavior of the network. By ALOHA transmission we mean the use of a shared channel which is randomly accessed by more than one user. In the simplest case users transit equal size packets, each using a data rate equal to the channel data rate several modes of operation are possible. The two simplest are: a non-slotted (asynchronous) mode in which users can access the channel at any time, and a slotted (synchronous) mode in which users can access the channel only at the beginning of a slot of time duration equal to a packet transmission time. In the latter case, a form of synchronization is required since each user must determine the beginning time of each slot. The following theoretical results assume that, if two or more packets overlap, none is correctly received, and each must be retransmitted. Such a system is called a system without capture.

The simplest analytic results assume that there are an infinite number of users and that the point process of packet origination and the point process of packet originations plus retransmissions are Poisson with mean S and G, respectively; constant transmission time T for each packet is also assumed. Then, if a packet begins at some random time, the probability that it is correctly received (no overlapping, collision or conflict) in the nonslotted case is e^{-2GT}. The reception rate, equal to the origination rate (assuming that colliding packets are retransmitted until correctly received) is $S = Ge^{-2GT}$. The effective channel utilization is $ST = GTe^{-2GT}$, and the maximum utilization is $\mathrm{Max}(ST) = 1/2e$. For the slotted case, the probability of collision is e^{-GT} which leads to $1/e$ as the maximum utilization. GT, the channel traffic is equal to 1/2 and 1 at a maximum effective utilization for the non-slotted and slotted case, respectively [1].

In the original ALOHA system, implemented at the University of Hawaii [2], a central station communicates with several remote sites. The system contains two channels—one for station-to-site traffic and the second for site-to-station traffic. This has several advantages for the ALOHA system. First, the station broadcasts continuously to furnish synchronization between all sites. Second, station-to-site traffic is coordinated by the station so that messages from the station do not collide with one another. Thus, if the traffic from the station has a separate channel from the reverse traffic, retransmissions are substantially reduced. Allocating separate channels for inbound and outbound station traffic is not as attractive when repeaters and multiple stations are introduced. This channel allocation problem is presently under investigation. Channel improvements also appear to be possible by using Spread Spectrum Coding, which offers the possibility of time capture. Competing packets arriving during the transmission time of the first may be ignored if their signal strength is not too great. When the transmitters are widely distributed, geometric or power capture is also possible [46]. With or without spread spectrum a competing signal which is much weaker (further away) than the desired signal will not interfere. Both types of capture can give rise to performance superior to that predicted by the simple unslotted ALOHA model. However, capture biases against more distant transmitters since the probability of a successful transmission to the station decreases as the distance from the station increases. Hence, it results in the increase in the number of retransmissions and consequently in the delay.

Chapter 11
PACKET RADIO NETWORK TOPOLOGY

11.1 General Considerations

Many factors affect the location of repeaters and stations. Simple consideration of repeaters as area covers and stations as traffic covers neglects interactions between the two types of devices.

Factors affecting the location of repeaters and stations in addition to range and traffic are:

a. **Logistics:** Some locations for repeaters may be preferable to others because of greater accessibility or more readily available power, eliminating the need for batteries (e.g., on telephone poles or near power lines).

b. **Reliability and Redundancy:** For many reasons, redundant repeaters and stations will be required. Since repeaters in remote areas will operate on batteries, it will be necessary to have sufficient redundancy so they need not be replaced immediately. Stations and repeaters will have intermittent and catastrophic failures for which backup is required. Extra repeaters are needed when line of sight to the primary repeater is locally blocked.

When a single channel is operated in an unslotted ALOHA random access mode, no more than 1/2e of the bandwidth can be effectively utilized, as discussed in the previous section. However, additional traffic is generated by repeaters, and conflicts created by transmissions between adjacent stations. Some sources of retransmissions are:

a. For reliability, several repeaters or stations must be within range of each terminal. If the repeaters retransmit every packet they receive, one message can generate an exponentially growing number of relayed messages. To prevent one message from saturating the network, traffic control is required. *The discipline chosen and its efficiency will probably be the single most important system factor affecting system performance.* Two types of undesirable routing through the repeaters can occur. First a message can circulate endlessly among the same group of repeaters if not controlled. Second, even if no message is propagated endlessly, a message can be propagated to a geometrically increasing number of new repeaters in a large network.

b. For system reliability, more than one station must be able to transmit via repeaters to each terminal. Thus, there can be conflicts between adjacent stations which reduces the useable bandwidth and also introduces coordination and routing problems.

c. In general, there will be many routes between any given terminal and any given station. Consequently, more conflicts can result than would be the case if the terminals communicated with a station.

11.2 Device Location

To provide line-of-sight coverage of an area where mobile terminals or fixed terminals are transmitting by radio from unspecified locations we must locate *repeaters* so that any such terminal will be in line-of-sight of repeaters and that there be reliable connections between every pair of terminals (and repeaters). More precisely, we wish to minimize the installation cost and maintenance cost of the repeaters subject to a constraint on the reliability of service.

In general, determining if line-of-sight microwave transmission between two points is possible involves taking into account many factors including wave-length (Fresnel zones), weather conditions (effective earth radius), antenna design, height, topography, etc. Nevertheless there are methods for making such calculations [42]. In this section, we describe methods for using the results of these determinations to choose good locations for the repeaters.

It is impractical to consider all possible locations of repeaters and terminals, which theoretically are infinite in number. We limit outselves to a finite set R of possible repeater locations and a finite set T of possible terminal locations. How the set R and T are chosen will be of great computational importance and will probably be chosen adaptively. But for the time being, we assume R and T known and fixed.

The principal and immediate interest is in an appropriate mathematical model of the situation and some indications on how to solve the problem. The first problem is the proper choice of reliability measure or grade of service. We assume that the radio network is for local distribution-collection of terminal traffic with rates small compared to the channel capacity so that throughput capacity is not a constraint. That is, if any path through the network exists for a given pair of terminals we assume there is sufficient capacity for traffic between them. Possible measures of network reliability that have proved useful in the analysis of communication network [55] are the probability that all terminal pairs can communicate and the average fraction of terminal pairs which can communicate. However, for network *synthesis* as distinguished from *analysis* these measures appear too difficult both from computational and data collection points of view. This suggests the "deterministic" requirement that there exist k node disjoint paths between every terminal pair. This guarantees that at least k repeaters or line-of-sight links must fail before any terminal pair is disconnected. Let the cost of a repeater at location $r \in R$ be $c(r)$ and $c(R^{\circ}) = \Sigma \{ c(r) \mid r \in R^{\circ} \}$ where $R^{\circ} \subset R$. Then, we can formulate:

Problem I

Find $R^* \subset R$ minimizing $c(R^*)$ subject to the constraint that for all $t \in T$ and $r \in R^*$ there exist k node disjoint paths from t to r.

One might demand only that there be k node disjoint paths between every pair of terminals instead of between each terminal-repeater pair, but we are assuming that communication always takes place through a "station" which could be any of the repeaters. The analysis of the terminal to terminal model is similar in any case.

The constraint can be broken into two parts:

a. k-fold set covering: The repeaters must be located so that at least k of them are in line-of-sight with each terminal, and

b. k redundancy: Between each pair of *repeaters* there must be k node disjoint paths.

Because the repeaters will have substantially greater range than terminals, the first aspect of the constraint will ordinarily be dominating. Moreover, it can be shown that the problem of minimizing costs of repeater locations subject to k-cover constraints is mathematically equivalent to 1-covering.

The 1-cover problem is the classical set covering problem. Extensive research has been and is being done on this problem, but there is good evidence—empirical [26] and theoretical—that the problem is intrinsically difficult.

Given the limited success to be expected from exact algorithms in solving large scale problems, we have been led to consider heuristic methods to find good solutions to the k-cover (of terminals by repeaters) problem which is typically large scale. It is intuitively appealing to consider a *terminal* as particularly critical if it is adjacent to few repeaters. (In the extreme cases, if a terminal has fewer than k adjacent repeaters, the problem is infeasible and if it has exactly k adjacent repeaters, all of them must be chosen for any feasible solution). Similarly, a *repeater* is desirable if it is adjacent to a large number of terminals, especially if the terminals are highly critical. The heuristic algorithms systemize these intuitive notions in the search of a "good" solution.

The size of test problems solved varies from problems with as few as 5 repeaters and 5 terminals to problems with as many as 400 repeaters and 400 terminals. Roughly speaking, the computation time was directly proportional to the size of the incidence matrix and the cover multiple required. The computer used is a PDP-10 (time sharing). The larger problems (400 repeaters, 400 terminals, 2-cover) were solved in 70 sec or less. The time, as may be expected, is dependent on the density of 1's in the incidence matrix. Thus, the maximum time recorded arose from terminals–repeaters configuration where each repeater covers many repeaters. The running time is of the order of $|T| \times |R|^2$ where $|T|$ and $|R|$ are the number of terminals and repeaters respectively.

We ran a number of problems with the heuristic code and for comparison with the Ophelie mixed integer programming code running on a CDC 6600 computer. The Ophelie code uses the branch-and-bound method. In the case of very simple problems (8 repeaters, 9 terminals, 2-cover) there was essentially no difference in running time (presumably most of the time, less than .5 sec, was spent in setting up the problem). Running experience with the Steiner triples' problem described in the next section, yields a ratio of 500 to 1 between the Ophelie time and the heuristic code time when solving the smaller problem A_{27} (117 terminals, 30 repeaters, 1-cover, and no comparison is available for the larger problem A_{45} (330 terminals, 45 repeaters) since for example, the MPSX code failed to reach a solution in more than one half hour on an IBM 360-91.*

Comparison in running time is naturally not completely valid, since most of the computation time in the Ophelie code can be spent just checking if a given solution is optimal. The heuristic method does not try to check the optimality of its solution However, in general, results with the heuristic code have been extremely good. When the heuristic solution deviated from the optimal solution, the problem usually involved numerous tries for the maximum $\omega_i^{(\ell)} \ell = 1, 2, 3, 4$ such as in the Steiner triples' problems. In all problems that were generated to resemble the packet radio terminal-repeater problem, the heuristic algorithm reached the optimal solution (in those problems for which we are able to determine the optimal solution).

[23] report on two covering problems which they characterize as computationally difficult. Each problem is defined by the incidence matrix of a Steiner triple system. The first problem, labelled A_{27} is a 1-cover problem with 117 terminals and 30 repeaters. The second problem, labelled A_{45}, has 330 terminals and 45 repeaters and is also a 1-cover problem. Data for both problems can be found on pages 9 and 10 of [23]. The problems are considered to be difficult because of the large number of verifications (branching in branch-and-bound, costs in cutting methods) required to establish that a given solution is in fact optimal.

The heuristics developed are dependent on the order in which the repeaters are presented to the algorithm. One hundred random permutations of the repeater ordering was tried for each of the two Steiner problems. In each case, the heuristic obtained the optimal solution for the smaller problem and for the larger problem, three solutions out of the 100 equaled the conjectured optimal solution of 30 repeaters. The remaining 97 were within

*Private Communication, [23].

3% of the optimals. On the other hand, we have constructed other artificial problems on which the heuristic performs abysmally.

Another test problem used to compare various techniques to solve the k-covering problem was generated by using real data obtained from a topographical map for the region of Palo Alto. This part of the U.S. was selected because it contained many interesting topographical attributes: a flat terrain (salt flats, the region surrounding the Bayshore Freeway), an urban center (Palo Alto and neighboring communities) on slightly sloping terrain and finally a hilly region (with valleys, small plateaus, etc.). Moreover, at this time, it appears that a reduced scale experiment of a packet radio network will be installed in the Palo Alto area.

LOS Computation

To determine if a terminal at location j can be seen from a repeater at location k, we proceeded as follows. It was assumed that if no particular high construction (building, water tower, etc.) was available to install the repeater's antenna, it would be installed at 30 feet above the ground level (making use of a tree, telephone pole, etc.). The terminals were assumed to be 5 feet above ground level. The points were said to be in LOS if the first *Fresnel Zone* associated with transmission between these two points was free of any obstacle (see Figure 11.1).

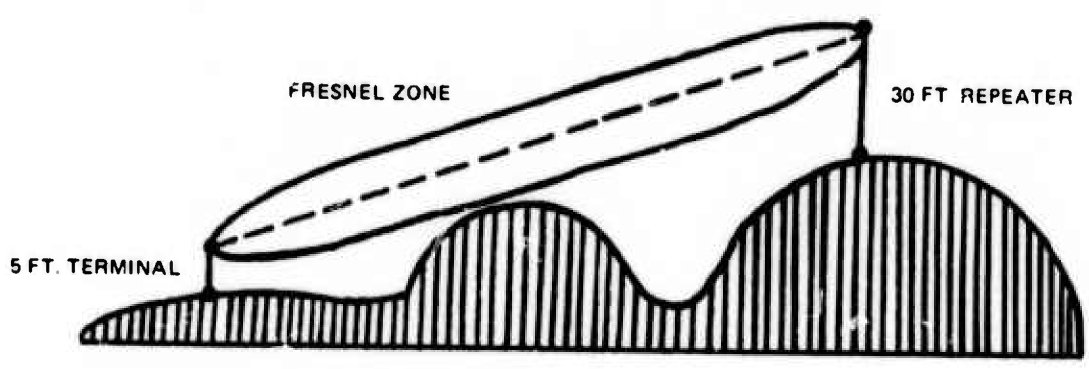


Figure 11.1: Relationship Between Terrain and Antenna Heights for Fresnel Zone

To compute the Fresnel Zones, we assumed that transmission would occur at 1500 MHz corresponding to a wave length $\lambda = .2$m (7.87 in.).

The problem was solved by the heuristic algorithm and by Ophelie. (A rapid analysis of the terminal-repeater adjacency matrix shows that none of the optimal solutions would have been generated if one had used the more simplistic approach of selecting the repeater with highest adjacency degree. Such a selection yields quite different answers requiring a larger number of repeaters).

The optimal solution requires the installation of 14 repeaters (different runs with the heuristic showed that there were, in fact, a number of optimal solutions with 14 repeaters). The total running time for Ophelie was approximately 12 CPU sec excluding set up time. The SETCOV required 3 sec to produce a solution. The relative success of the Ophelie code must, at least in part, be attributed to the fact that the linear programming solution (which is used to initiate the branch-and-bound part of the code) is actually the optimal solution. The input contour configuration and repeater location solution for this sample problem are shown in Figures 11.2 and 11.3, respectively.

11.3 Network Topological Reconfiguration

From the general topological considerations, it is apparent that the routing and flow control algorithms will be the main factor which will determine the efficiency of the Packet Radio System. However, there are two contradictory requirements; reliability considerations advocate that every repeater should be able to transmit to several repeaters; on the other hand, efficiency consideration suggest that one repeater should receive and relay the packet, preferably the repeater along the shortest path to the destination. A sensible solution is to assign to the set of repeaters a structure which will transform the broadcast network to a point-to-point network for routing purposes. The problem is that the connectivity of devices is changing, and therefore, it is necessary to develop algorithms for dynamically changing the network structure (reconfiguration) under certain conditions. Examples of a changing topology are when the network is mobile (e.g., a Packet Radio System for a fleet of ships), drainage of battery power of repeaters placed in inaccessible environments, or when repeaters fail to operate.

In [41], we propose algorithms for dynamically changing the network configuration. It is assumed that every repeater and station have a fixed ID, and that there is a simple routing algorithm however inefficient, which is independent of any network structure. The process contains three steps:

STEP I

Mapping the network connectivity. This is obtained by a process in which stations transmit packets to repeaters, requesting each to respond with a trace packet into which every repeater along the path adds its fixed ID.

STEP II

Determining network structure. The connectivity information obtained above is used to obtain a network structure which has several properties; for example, it enables every packet to be routed along the shortest path (minimum number of hops); it determines the repeaters which are not needed for relaying packets and which should be temporarily disabled.

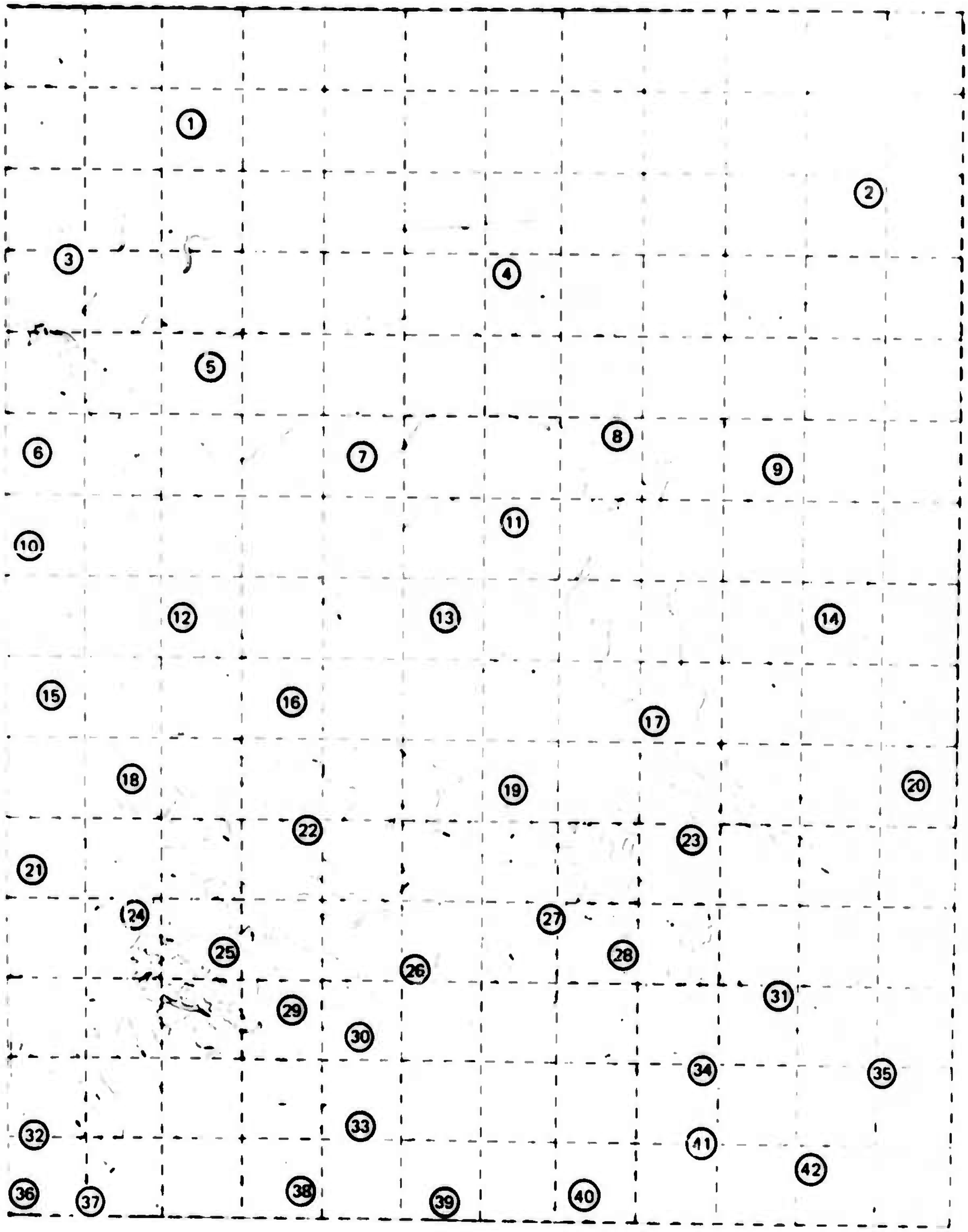


Figure 11.2: Contour Map and Available Repeater Locations for Example

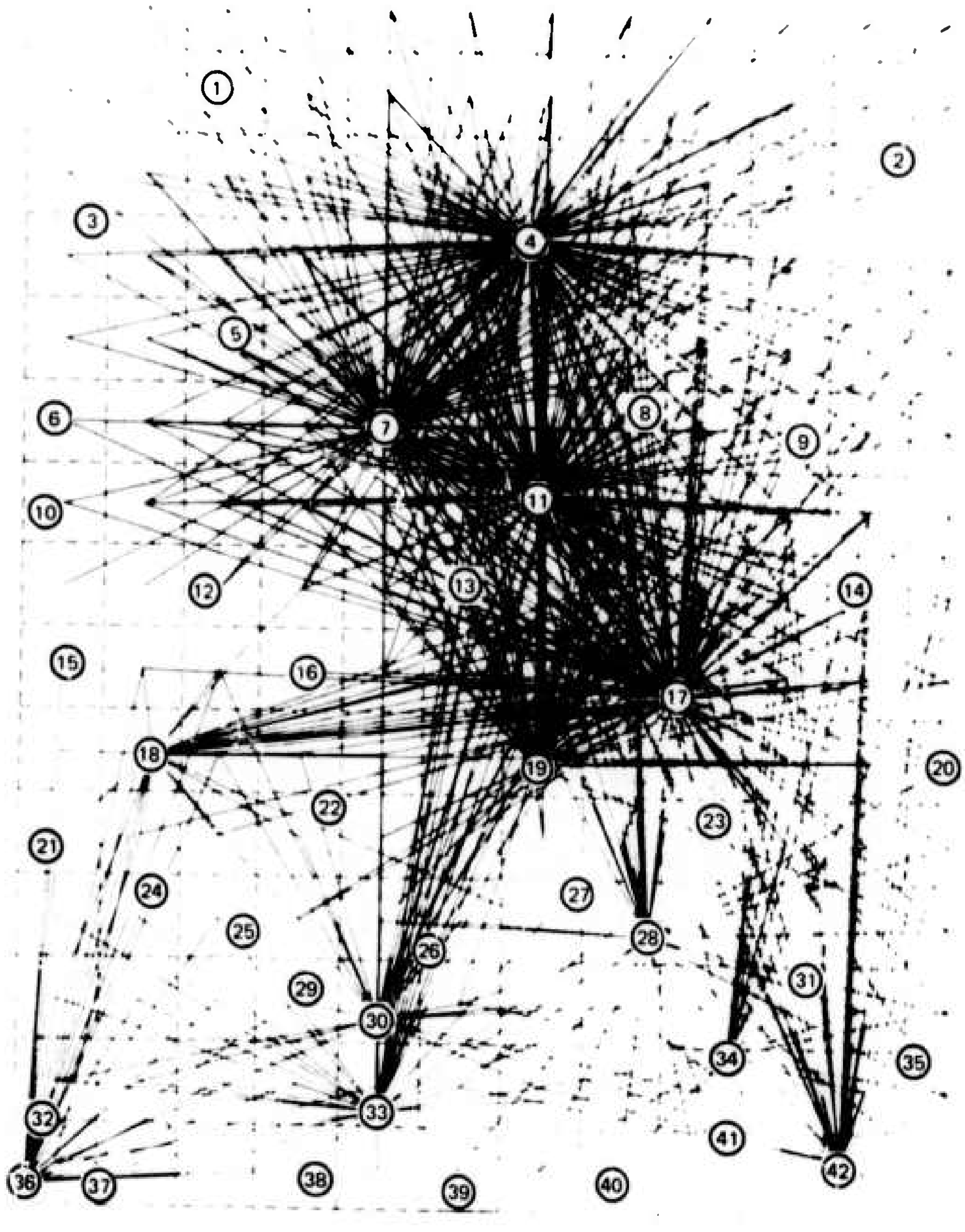


Figure 11.3: Repeater Covering for Example

STEP III

In this step, the stations transmit the structure information to repeaters and test each path in both directions.

Chapter 12
PACKET RADIO SYSTEM CHANNEL CONFIGURATION

12.1 Split versus Common Channel

Apart from the suitability for mobile terminals, random access schemes offer an attractive alternative to fixed assignment of channel capacity (FDM, TDMA) for applications characterized by traffic of a bursty nature. (That is, when the traffic requirements of users can be characterized as having a high peak to average data rate.) This is because at any given time, the capacity assigned to non active users is not utilized, whereas the active users experience relatively long delays due to the low data rate available to each.

We pursue this same argument one step further and investigate for the packet radio system whether we should have two channels, one for transmission from terminals to stations and the second in the reverse direction; or alternatively whether we should dynamically share the total capacity (common channel). This problem was investigated for a single hop network in which n stations communicate with an infinite number of terminals using the slotted ALOHA random access scheme [28]. In the model it is assumed that all stations and terminals are within an effective transmission range of each other, that the processes of packet originations and packet originations plus retransmissions are Poisson, and that there is a ratio a of the rate of packets which originates from stations to the rate which originates from terminals.

Figure 12.1 shows the comparison of the maximum effective utilization of the two configurations as a function of a with the number of stations, n, as a parameter. The subscripts s and c denote the split (into equal parts) and common configurations, respectively. The conclusion from this is that if the ratio a is not known or if it varies, it is preferable to share dynamically the total capacity. Figure 12.2 shows an example of the average delay of a packet in the system (weighted average of packet in the two directions) as a function of the total throughput, for the case $a = 10$. The difference in the packet transmission time (slot) due to the difference in the data rates of the two configurations has been taken into account. The superiority of the common channel configuration in this case ($a = 10$) is clearly demonstrated.

12.2 Directional Antennas and Multiple Transmitters

Another problem related to channel configuration is the possible use of directional antennas by repeaters and/or stations and the advantage (if any) of using multiple directional

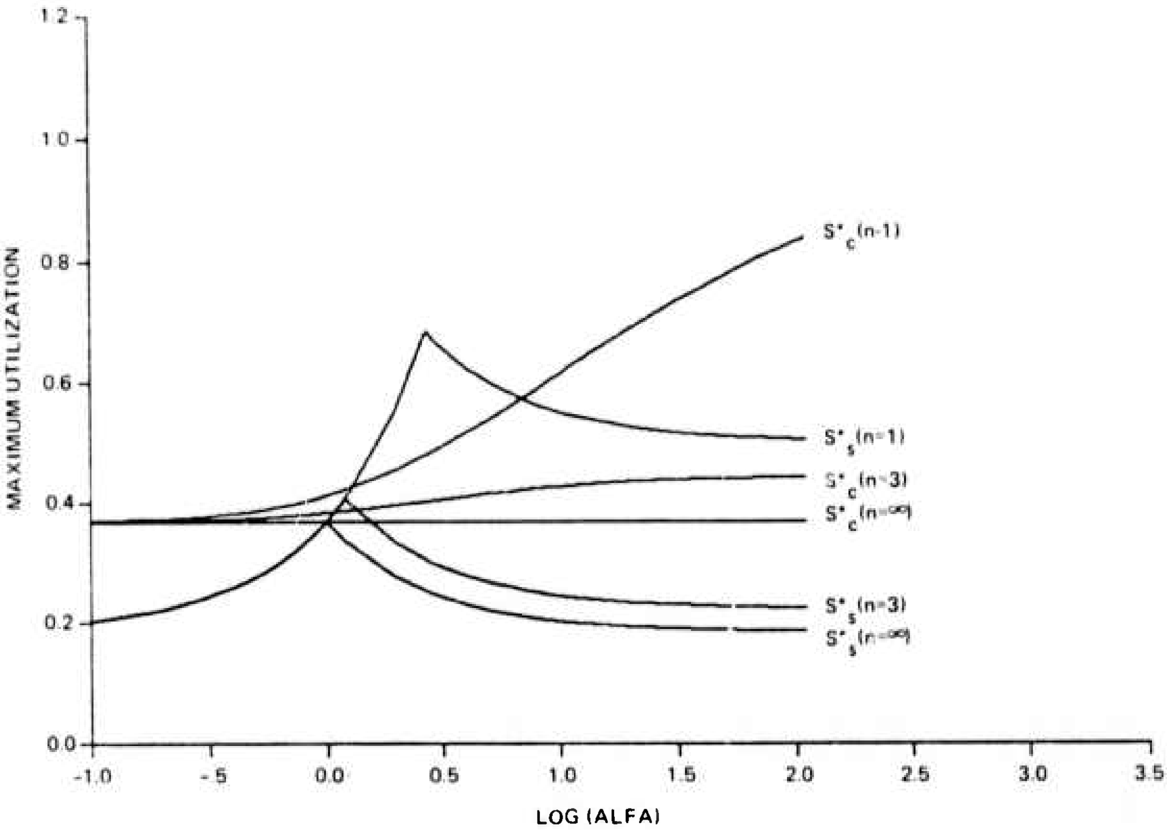


Figure 12.1: Maximum Utilization vs. Split and Combined Channel Parameters

transmitters. This problem was investigated for a 2-hop and single station packet-radio network [29]. The investigation was done assuming separate channels from station to terminals and from terminals to station, and for the slotted ALOHA random access scheme.

12.2.1 Transmission From Terminals To Station

Consider a 2-hop system with m repeaters and a single station as shown in Figure 12.3. The traffic originates from terminals and is destined to the station. A terminal transmits its packets to a repeater (hop 1), which in turn transmits the packets to the station (hop 2). The transmission protocol is as follows: when a packet becomes ready for transmission, it is transmitted into the next slot; the device then times out waiting for an ack, and if one is not received the packet is retransmitted at a future random slot.

We use the following assumptions. The combined process of packet originations and packet retransmissions, from each set of terminals to a repeater, is Poisson. The probabilities of transmission by a repeater into different slots are independent. The probability of transmission by two or more repeaters into a randomly chosen slot are mutually independent; and the probability of transmission into a random slot by a terminal and by a repeater are

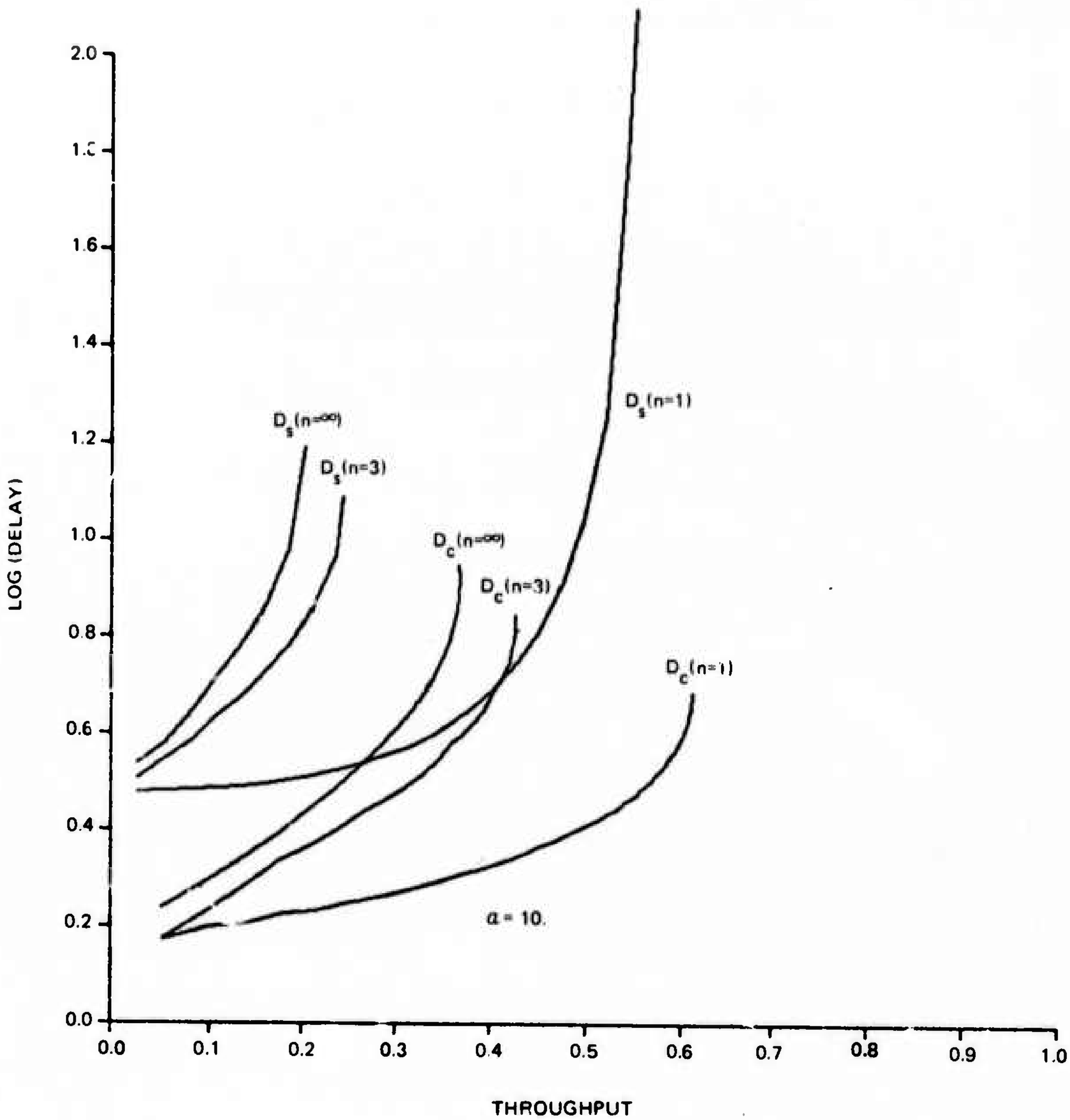


Figure 12.2: Delay vs. Throughput, a = 2.5

independent. Furthermore, we assume that the terminal transmission range is short, so that it can reach only one repeater. On the other hand, the transmission from a repeater to the station can interface with the transmission of terminals to l-1 other repeaters; $1 \leq 1 \leq m$.

The effect of directional antennas at repeaters is that the transmission from repeaters to the station is directed towards the station and does not interfere with the transmission of terminals to other repeaters. Thus, it is the special case with l = 1. We notice, however, that directional antennas do not increase the capacity of the hop from repeaters to station

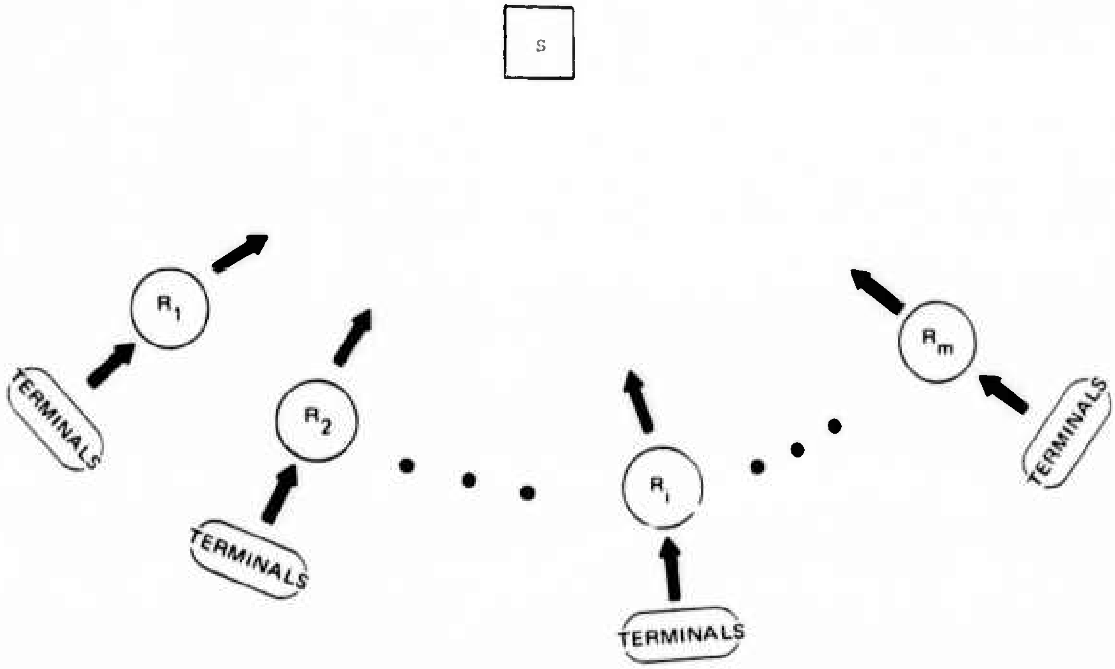


Figure 12.3: Transmission from Terminals to Station

because all antennas are directed towards the same physical location where the station is placed and where the conflicts may occur.

Figure 12.4 shows the capacity of the system as a function of the number of repeaters, m, for I = m and I = 1, which is equivalent to omnidirectional and directional antennas respectively. One can see that there is a significant gain in capacity when using directional antennas only when m = 2, and a small gain for m = 3; for m $\geq$ 4 the capacity of the system does not increase.

As far as the number of repeaters is concerned, one can see 2 or 3 repeaters would be a good design; and additional repeaters that may be added because of other considerations (such as area coverage) will result in a reduction in the system capacity. Another problem investigated is the critical hop. That is, when the capacity of the system is reached, it is due to the saturation of the hop from terminals to repeaters or that from repeaters to the station. The results demonstrate that when the number of repeaters, m, is small the critical hop is from terminals to repeaters, whereas when m is large the critical hop is from repeaters to station. The exact number at which the change occurs depends on the interference parameter I.

12.2.2 Transmission From Station To Terminals

In this section, we consider the second channel which is used for transmission from the station to terminals via repeaters. It is assumed that the effective transmission range of

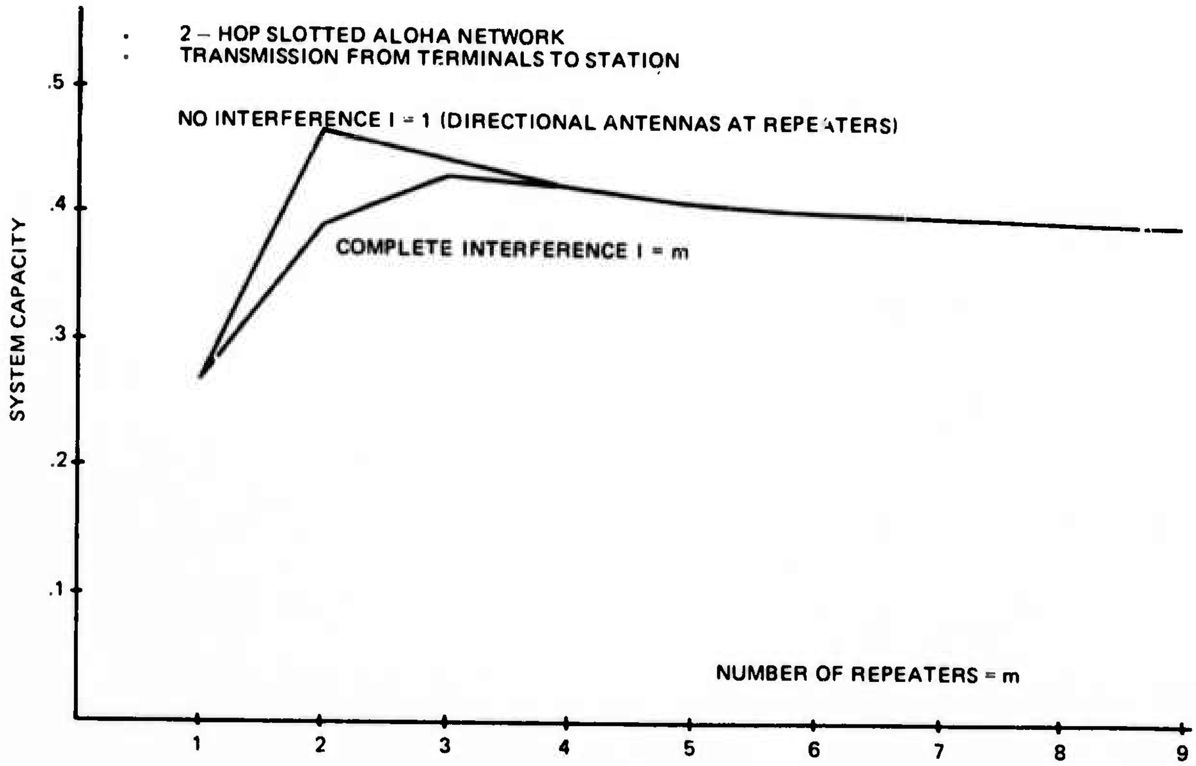


Figure 12.4: Network Throughput vs. Number of Repeaters: Directional and Non-Directional Antennas

the station is such that it interferes with the transmission from repeaters to terminals. However, we assume that terminals are not designed to directly receive from the station. We use the same assumptions as in the previous section. A transmission from the station to R_i can be interfered with by transmissions from the I repeaters in the interfering set of R_i when these repeaters transmit to their terminals (T's). A transmission from R_i to T can be interfered by a transmission from the station to any repeater or by the I – 1, excluding R_i, repeaters in the interfering set of R_i. For consistency with the interference model in the previous section we assume the same energy-per-bit-to-noise-density for detection with equal error rates, by the repeaters and by the terminal and that the repeater uses a higher transmitter power than terminals.

Figure 12.5 shows the capacity of the system as a function of m for I = 1 and I = m, both for an omnidirectional and directional antenna from the station to repeaters. Further investigations for this case were performed and the conclusions follow.

a. The interference of the station with the transmission of repeaters to terminals significantly reduces the system capacity. Thus, if possible, it is important to enable terminals to receive such transmissions directly, without retransmission by the repeater.

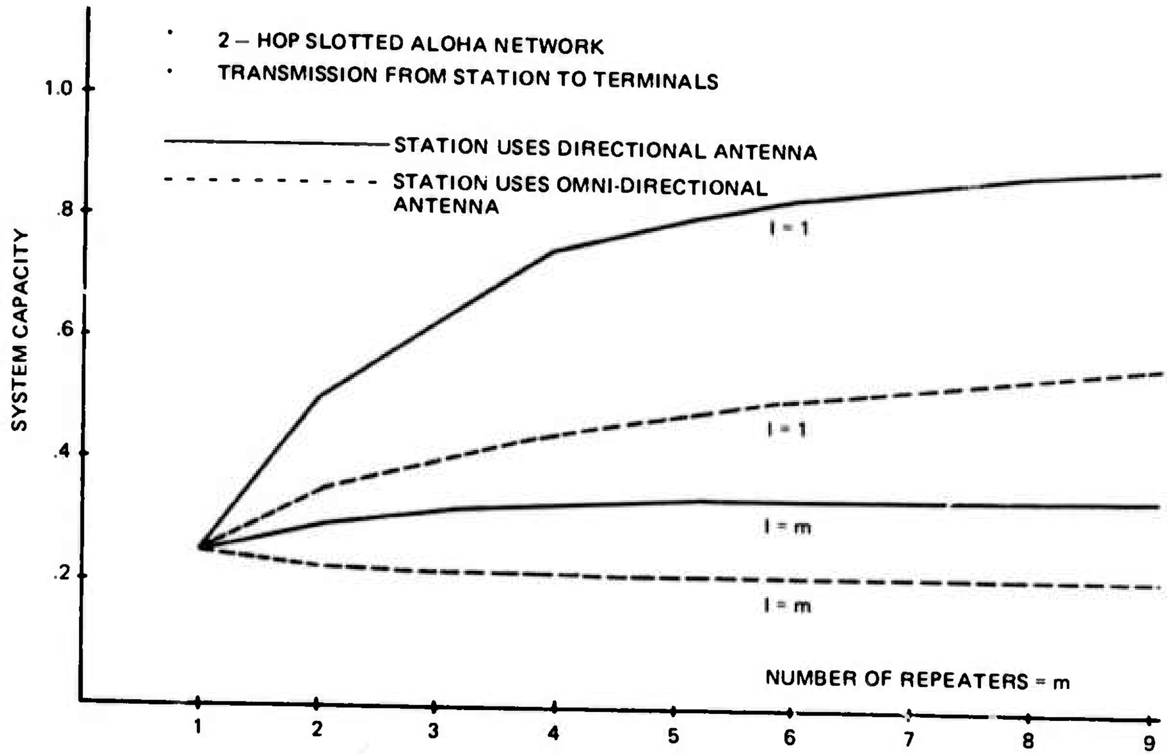


Figure 12.5: Number of Repeaters vs. Capacity

b. The system capacity is reduced substantially when the interference level between repeaters is increased. Note that this is not the case when transmitting to the station. Consequently, it is important to reduce the interference factor by a mechanism such as adaptive power.

c. A directional antenna at the station significantly increases the system capacity when the interference level between repeaters is low to moderate. This is not the case when the interference level is high, since the throughput on the hop from repeaters to terminals is limited due to this interference.

d. When the station has directional antennas, then multiple transmitters and antennas may further increase significantly system capacity. In this case one can obtain a capacity greater than 1.

Chapter 13
PACKET RADIO SYSTEM ROUTING AND ACKNOWLEDGEMENT CONSIDERATIONS

13.1 Routing Problems

Problems that arise in controlling traffic in a broadcast net include:

a. A packet transmitted can be received by many repeaters or stations or not be received by any.

b. Many copies of the same packet can circulate in the broadcast network.

c. Many copies of the same packet can enter the point-to-point network at different stations.

Indications of the consequences of not imposing a suitable flow control mechanism can be observed from idealized combinatorial models.

In these ideal models, the repeaters are located at corner points of an infinite square grid and time is broken into unit intervals, each slotted into segments. A packet transmitted by a repeater can be received only by its four nearest neighbors. If a packet is correctly received by a repeater, it is retransmitted within the next unit interval of time at a random time slot within the interval. Suppose now that *a single packet* originates at the origin and that the transmission plus the propagation time falls within one unit interval of time. Then after n intervals of time:

a. The number of repeaters which receive the packet for the first time, B(n), is:

$$B(n) = 4n, \; n > 1 \quad B(0) = 1$$

b. The number of repeaters through which the packet passed, A(n), is:

$$A(n) = \sum_{j=0}^{n} B(j) = 2n^2 + 2n + 1, \; n \geq 0$$

c. if we assume that a repeater can receive and relay a large number of packets within the same time interval, the number of copies of the same packet received by a repeater at coordinates (d,j) after d + 2k units of time is:

$$N_j^d\ (d =+ 2k) = \binom{d+2k}{k+j}\binom{d+2k}{k} \xrightarrow{\text{for large } k} 2^{4k}$$

where d is the number of units of time that the packet requires to arrive from the origin to the repeater, and j is the horizontal number of units.

Unless adequate steps are taken, the explosive proliferation of redundant packets will severely limit the capacity of the system. One can now recognize two somewhat distinct routing and control problems:

a. To ensure that a packet originating from a terminal arrives at a station, preferably using the most efficient (shortest) path,

b. To suppress copies of the same packet from being indefinitely repeated in the network, either by being propagated in endless cycles of repeaters or by being propagated for a very long distance.

13.2 Proposed Routing Techniques

There are two key objectives in developing a routing procedure for the packet radio system. First, we must assure, with high probability, that a message launched into the net from an arbitrary point will reach its destination. Second, we must guarantee that *a large number* of messages will be able to be transmitted through the network with a relatively small time delay. The first goal may be thought of as a *connectivity* or *reliability* issue, while the second is an *efficiency* consideration.

13.2.1 Undirected Routing

A rudimentary, but workable, routing technique to achieve connectivity at low traffic levels can be simply constructed by using a maximum handover number [4] and saving unique identifiers of packets at each repeater for specified periods of time. The handover number is used to guarantee that any packet cannot be indefinitely propagated in the net. Each time a packet is transmitted in the net, a handover number in the header is incremented by one. When the handover number reaches an assigned maximum, the packet is no longer repeated and that copy of the packet is dropped from the net. Thus, the packet is "aged" each time it is repeated until it reaches its destination or is dropped because of excessive age.

If the maximum handover number is set large, extensive artificial traffic may be generated in areas where there is a high density of repeaters. On the other hand, if it is set small, packets from remote areas may never arrive at stations. This problem can be resolved as follows: We assume that every repeater can calculate its approximate distance in numbers of hops to stations by observing response packets. The first repeater which received the packet from a terminal sets the maximum handover number based on its calculated distance from the station. The number is then decremented by one each time it is relayed through any other repeater. The packet is dropped when the number reduces to zero. When a station transmits a packet, it will set the maximum handover number by "knowing" the approximate radius in "repeaters" in its region.

Even if a packet is dropped after a large number of transmissions, local controls are needed to prevent packets from being successively "bounced" between two or a small number of repeaters which repeat everything they correctly receive. (Such a phenomena is called "cycling" or "looping.") A simple mechanism to prevent this occurrence is for repeaters to store for a fixed period of time entire packets, headers, or even a field within the header that uniquely identifies a packet. A repeater would then compare the identifier of any received packet against the identifiers in storage at the repeater. If a match occurred, the associated packet would not be repeated.

The time allotted for storage of any packet identifier would depend on the amount of available storage at a repeater and the number of bits required to uniquely identify the packet. For example, more than 4K packets could be uniquely identified with 12 bit words. Thus, 4K of storage could contain identifiers for more than 300 packets. With a 500 Kbps repeater to repeater common channel for broadcast and receive and 1,000 bit packets, this would be sufficient storage for over 1.5 seconds of transmission if the channel were used at full rate. Assuming a single hop would require about 20 milliseconds of transmission and retransmission time, a maximum hop number of 20 would guarantee that any packet would be dropped from the system because of an excessive number of retransmissions long before it could return to a previously used repeater not containing the packet identifier.

The combination of loop prevention and packet aging with otherwise indiscriminate repetition of packets by repeaters will enable a packet to travel, on every available path, a maximum distance away from its origin equal to its original handover number. Thus, if the maximum handover number is larger than the minimum number of hops between the terminal and the nearest station, a packet accepted into the net should reach its destination. Unfortunately, with this scheme, copies of the packet will also reach many other points, with each repetition occupying valuable channel capacity. However, if those packets for which adequate capacity is not available are prevented from entering the net, the network will appear highly reliable to accepted packets.

The above routing scheme is an *undirected*, completely distributed procedure. Each repeater is in total control of packets sent to it, and the stations play no active part in the system's routing decisions. (They must still play a role in flow control.) In the above procedure, no advantage is taken of the fact that most traffic is destined for a station, either as a terminus or as an intermediate point for communication with elements of a different network. Also, the superior speed and memory space of the station is ignored. For efficiency, one is therefore led to investigate *directed* (hierarchical) routing procedures.

13.2.2 Directed Routing

A directed routing procedure utilizes the stations to periodically structure the network for efficient flow paths. Stations periodically transmit routing packets called *labels* to repeaters to form, functionally, a hierarchical point-to-point network as shown in Figure 13.1. Each label includes the following information:

a. A specific address of the repeater for routing purposes

b. The minimum number of hops to the nearest station

c. The specific addresses of *all* repeaters on a shortest path to the station (In particular, the label contains the address of the repeater to which a packet should preferably be transmitted when destined to the station.)

When relaying a packet to its destination, the repeater addresses the packet to the next repeater along the preferred path. Only this addressed repeater will repeat the packet and only when this mechanism fails will other repeaters relay the message.

For simplicity, we describe routing for the case of a single station network. A label of repeater R_i of hierarchy level j will be denoted by L_{ij}; i,j>1. The station will have the label L_{11}. L°_{ij} will denote the label of the repeater which is the "nearest available" to the communicating terminal.

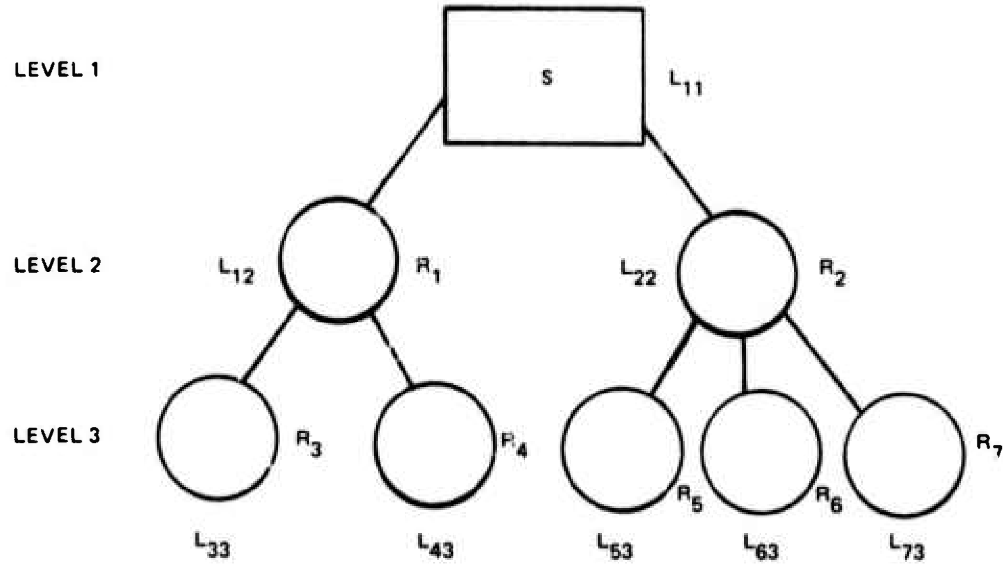


Figure 13.1: Hierarchical Labeling For Directed Routing Algorithms

A label is composed of H subfields, where H is the maximum number of hierarchy levels (H-1 is the maximum number of hops on the shortest path between any repeater and the station). Every subfield has three possible entries, blank (BLK), a serial number (SER), or ALL. L_{ij} has j entries of SER's and (H-j) BLK's as shown in Figure 13.2.

1	2		j-1	j	j+1		H
SER	SER	....	SER	SER	BLK	....	BLK
j serial numbers					(H-j) blanks		

Figure 13.2: Definition of Packet Label

We say that L_{ij} "homes" on L_{kp}, $h(L_{ij}) = L_{kp}$, if p = j-1 and the first j-1 subfields of both are identical. If two repeaters at level j home on the same repeater, their label will differ only in the entry to subfield j.

As an example, if we use 3 bits per subfield, the labels of the station and the repeaters of the network shown in Figure 13.1 are as follows:

	Subfield 1	Subfield 2	Subfield 3
L_{11}	0 0 1	0 0 0	0 0 0
L_{12}	0 0 1	0 0 1	0 0 0
L_{22}	0 0 1	0 1 0	0 0 0
L_{33}	0 0 1	0 0 1	0 0 1
L_{43}	0 0 1	0 0 1	0 1 0
L_{53}	0 0 1	0 1 0	0 0 1
L_{63}	0 0 1	0 1 0	0 1 0
L_{73}	0 0 1	0 1 0	0 1 1

In this example, a subfield in which all bits are "0" is considered "blank." Note that all entries in Subfield 1 are the same since all repeaters home (eventually) on the same station.

The packet header, shown in Figure 13.3, includes the following information.

L_{kn}	L°_{ij}	OTHER HEADERS AND PACKET INFORMATION
TO	LABEL OF NEAREST REPEATER TO THE TERMINAL	

Figure 13.3: Routing Information Contained in Packet Header

L_{kn} is the label of the repeater to which the packet is currently addressed. The complete packet will *always* be transmitted to a *specific device;* other devices which may receive the packet will drop it. The shortest path from a terminal to the station consists of L°_{ij}, $h(L^\circ_{ij})$, $h(h(L^\circ_{ij}))$, up to L_{11}, in the given order, and in the reverse order when routing from station to terminal. When a specific repeater along the shortest path is not known (by the terminal) or not available, then the terminal or repeater (which has the packet) will transmit *only the header part* of the packet, trying to identify a specific repeater. In that case, the label L_{kn} will include some entries ALL. To see how the proposed routing technique would operate, we trace the sequence of steps performed when a terminal attempts to transmit a packet to the station.

When a previously silent terminal begins to communicate, it first identifies a repeater or a station in its area. It transmits only the header part of the packet with all entries in L_{kn} set to ALL. The header is addressed to all repeaters and stations that can hear the terminal. A device which correctly receives this header substitutes its label in the space L_{kn} and repeats the header. This particular L_{kn} is also L°_{kn} and will be used by the terminal to transmit all packets during this period of communication. If a terminal is stationary, it can store this label for future transmissions. L°_{kn} begins to transmit the complete packet along the shortest path to the station.

Suppose that L_{ij} along the shortest path is not successful in transmitting the packet to $h(L_{ij})$. Then L_{ij} begins the search stage of trying to identify another repeater. In the first step, it tries to identify a repeater which is in level $p \leq j-1$. This is done by using the label shown in Figure 13.4.

1	2	3		j-1	j	j+1		
SER	ALL	ALL	...	ALL	BLK	BLK	...	BLK

Figure 13.4: Label Used In Search Process

The header is addressed to all repeaters in levels 2 to j-1, which eventually home on L_{11}. If this step is not successful, in the second (last) step, L_{ij} tries to identify any available repeater by using a label in which the first entry is SER and all other entries are ALL. When a specific repeater is identified and receives the packet, it transmits the packet on the shortest path from its location.

Note that if repeaters have sufficient storage, they can save alternative labels and thus reduce the necessity of searching for a specific repeater. Alternative solutions in which repeaters have multiple labels are also possible.

13.3 Acknowledgement Considerations

Acknowledgement procedures are necessary both as a guarantee that packets are not lost within the net and as a flow control mechanism to prevent retransmissions of packets from entering the net. Two types of acknowledgements are common in packet oriented systems:

a. Hop-by-Hop Acknowledgements (HBH Acks) are transmitted whenever a packet is received successfully by the next node on the transmission path.

b. End-to-End Acknowledgements (ETE Acks) are transmitted whenever a packet correctly reaches its final destination within the network.

In a point-to-point oriented network such as the ARPANET, HBH Acks are used to transfer responsibility (and thus open buffer space) for the packet from the transmitting node to the receiving node. This Ack insures prompt retransmission should parity errors or relay IMP buffer congestion occur. The ETE Ack serves as a flow regulator between source and destination and as a signal to the sending node that the final destination node has correctly received the message. Thus, the message may be dropped from storage at its origin.

Both types of Ack's serve to ensure message integrity and reliability. If there is a high probability of error free transmission per hop and the nodes have sufficient storage, the Hop-by-Hop scheme is not needed for the above purpose. Without an HBH Ack scheme, one would transmit the packet from its origin after a time out period expired. One introduced the HBH Ack to decrease the delay caused by retransmissions at the expense of added overhead for acknowledgements. In the ARPANET, this added overhead is kept small by "piggybacking" acknowledgements whenever possible on information packets flowing in the reverse direction. In the packet radio system, the overhead can be kept small by listening, whenever possible, for the next repetition of the packet on the common channel instead of generating a separate acknowledgement packet.

The value of an End-to-End acknowledgement is sufficiently great that it can be assumed present *a priori*. However, the additional use of a Hop-by-Hop acknowledgement is not as clear. Therefore, in this section, we examine the question of whether the ETE Ack is sufficient, or whether one needs a Hop-by-Hop (HBH) acknowledgement in addition. The problem is therefore whether an HBH Ack is superior to an ETE Ack with respect to throughput and delay, since the ETE Ack ensures message integrity. It is noted that the routing and flow control by devices in the network depend on the type of acknowledgement scheme used.

We consider a simple case where (n-1) repeaters separate the packet radio terminal from the destination station. Assuming that the terminal is at a distance of "one hop" from the first repeater, one obtains the n-hop system shown in Figure 13.5.

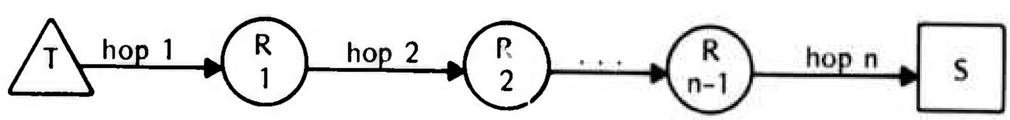


Figure 13.5: Multi-Hop Stream

A simple model is used to evaluate the total average delay that a packet encounters in the n-hop system when using HBH and ETE acknowledgement schemes. When the ETE acknowledgement scheme is used, every repeater transmits the packet a single time. If the packet does not reach the station, retransmission is originated by the terminal. The ETE acknowledgement is sent from the station. In the HBH scheme, repeaters store and retransmit the packet until positively acknowledged from the next repeater stage.

If, after a terminal (or a repeater in the HBH case) transmits the packet, an acknowledgement does not arrive within a specified period of time, it retransmits the packet. The waiting period is composed of the time for the acknowledgement to arrive when no conflicts occur plus a random time for avoiding repeater conflicts.

Two different schemes for ETE acknowledgement and one scheme for HBH acknowledgement have been studied. Curves for the total average delay as a function of the number of hops and the probability of successful transmission per hop are obtained. Two cases are considered: One in which the probability of success is constant along the path and another in which the probability of success decreases linearly as the packet approaches the station. Finally, channel utilizations are compared when using ALOHA [1] random access modes of operation.

It has been demonstrated that the HBH scheme is superior in terms of delay or channel utilization. This conclusion becomes significant when the number of hops increases or when the probability of successful transmission is low. For example, in a five hop system, if the probability of success per hop is 0.7, then the total average delay is 12.5 and 53 packet transmission times for the HBH and ETE acknowledgement schemes, respectively.

Chapter 14
RANGE, POWER, DATA RATE AND CAPACITY CONSIDERATIONS

14.1 Transmission Range and Network Interference

A variety of situations is possible concerning the range and interference patterns of devices. For example, with identical r.f. elements and similar antenna placements, Repeater-to-Repeater range is the same as Terminal-to-Repeater range. This, however, is not always a necessary limitation since repeaters can be placed on elevated areas and can have more power than terminals (especially hand held terminals). Thus, if repeaters are allocated for *area coverage of terminals*, the range will be higher than terminal range and higher network connectivity or device interference will result.

The problem which then arises is to determine the impact of this interference on system performance. Alternatively, one may seek to reduce repeater transmission power when transmitting in the repeater-station network. As an indication of the tradeoffs that occur, common channels and the single date rates (CCSDR) were simulated, one with High Interference CCSDR (HI), and the other with Low Interference CCSDR (LI). As a first step, the routing labels of the two systems were the same and are shown in Figure 14.2. The interference of the CCSDR (LI) system is shown in Figure 14.1 and the interference of the CCSDR (HI) system in Figure 14.3. (Figure 14.3 shows only the connectivity for two devices in the network.) A different label assignment for the high interference system is shown in Figure 14.4.

The results are shown in Figure 14.5 and Table 14.1. Figure 14.5 shows the throughput of the two systems as a function of time while Table 14.1 summarizes other measures of performance. The third row of Table 14.1 summarizes performance of the high interference system under an improved set of repeater labels. It is clear that the high interference system has better performance than the low interference system. The only measure of the low interference system which is better is terminal blocking which is a direct result of the low interference feature. In fact, CCSDR (LI) is saturated at the offered traffic rate. This can be seen from the fact that the throughput is decreasing as a function of time, the relatively high total loss, and the low station response.* The CCSDR (HI) with improved labels, compared in Table 14.1, has better performance than the other two systems. This indi-

*The average number of station response packets assumed for these studies is 2.0.

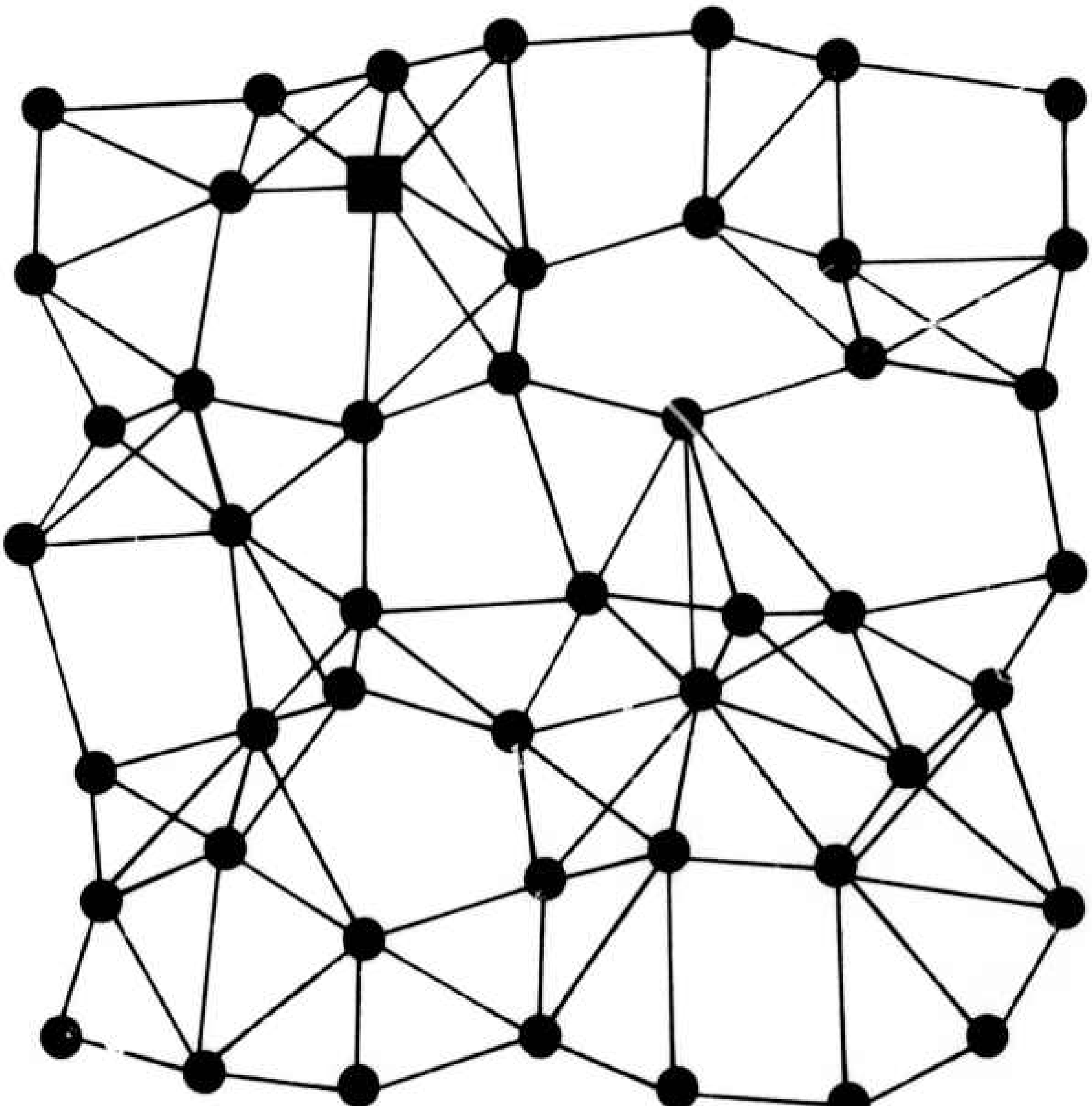


Figure 14.1: Connectivity of Repeaters and Stations

cates the importance of proper labeling. The experiments of this section demonstrate that it is preferable to use high transmitter power to obtain long repeater range, despite the network interference that results.

14.2 Single Versus Dual Data Signaling Rates Networks

The previous results demonstrate that better performance is obtained when repeaters and stations use high power to obtain long range despite the interference that results. We now examine the problem of whether repeaters and stations should use their fixed power budgets to obtain a long range with a low data rate channel or have a short range with a high data rate channel. The following systems were studied.

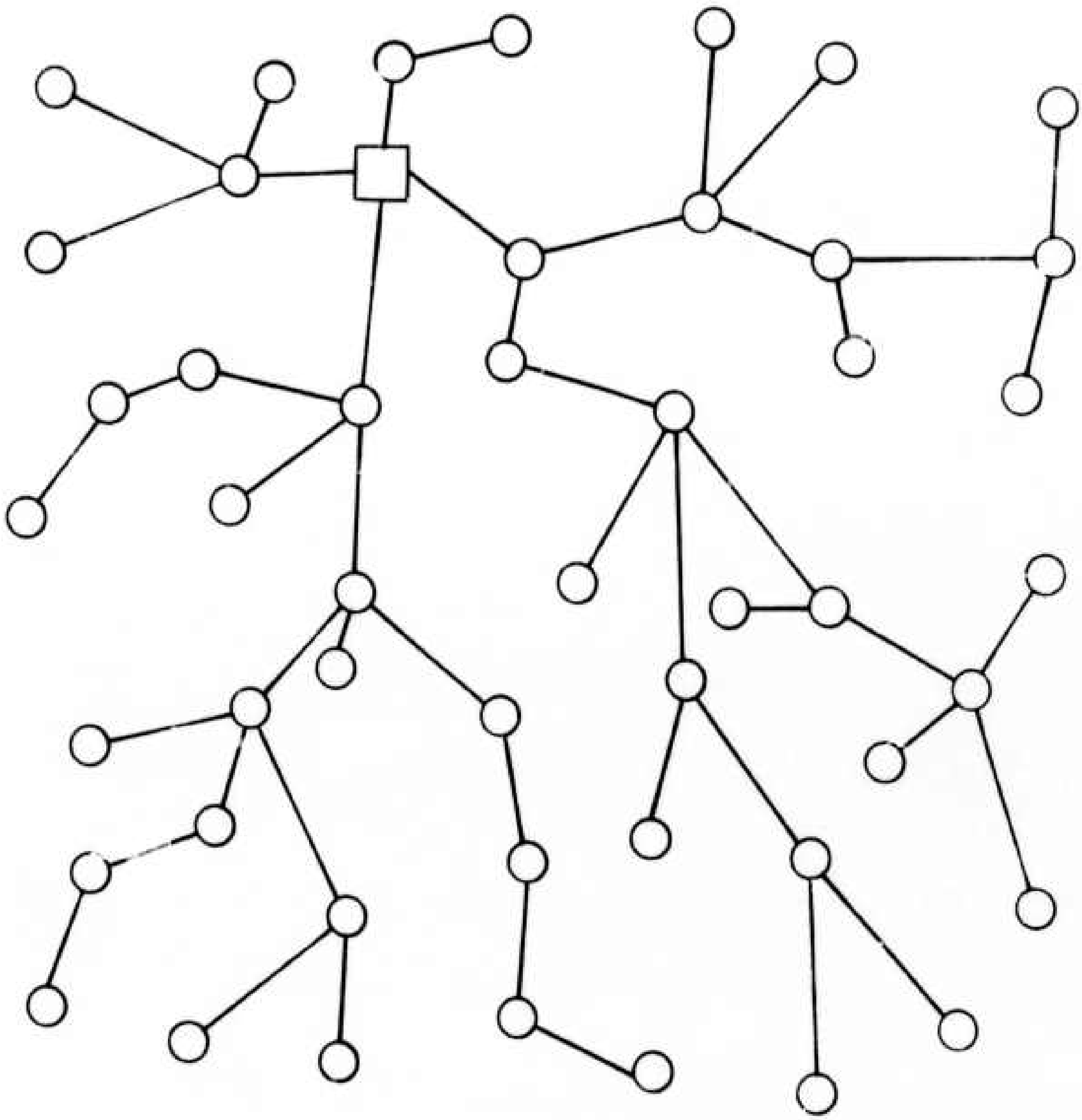


Figure 14.2: Hierarchical Labeling Scheme

- The CCSDR (HI) of the previous section with improved label_ to take advantage of the high range to improve the routing labels of repeaters and obtain fewer hierarchy levels which we denote by CCSDR. The routing labels used are shown in Figure 14.4, and the connectivity is shown in Figure 14.3.

- A Common Channel Two Data Rate (CCTDR) system with the routing labels as in Figure 14.2 and connectivity as in Figure 14.1.

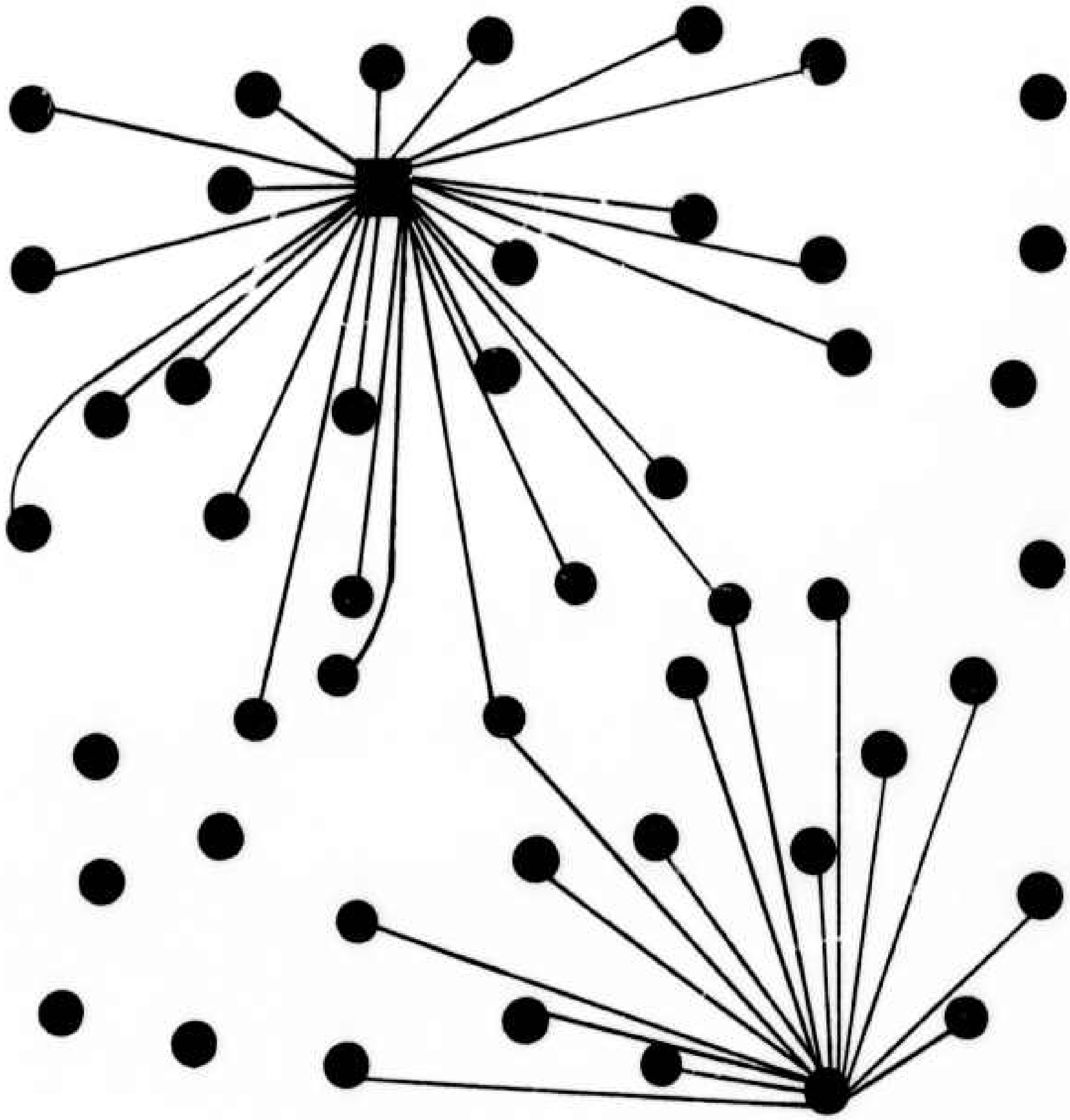


Figure 14.3: Interference of CCSDR (HI) System

In the CCTDR system, the terminal has a low data rate channel, the same rate as in the single data rate system, for communication with a repeater or station. Repeaters and stations have two data rates. The high data rate is used for communication in the repeater-station network. The two data rates use the same carrier frequency so that only one can be used at a time.

The two systems are tested with offered rates of 13% and 25%.* The throughput as a function of time for the two runs is shown in Figures 14.6 and 14.7, respectively; and

*In the simulation runs we used the inverse square law for the relation between data rate and distance, rather than the result in [9]; this, however, favors CCSDR.

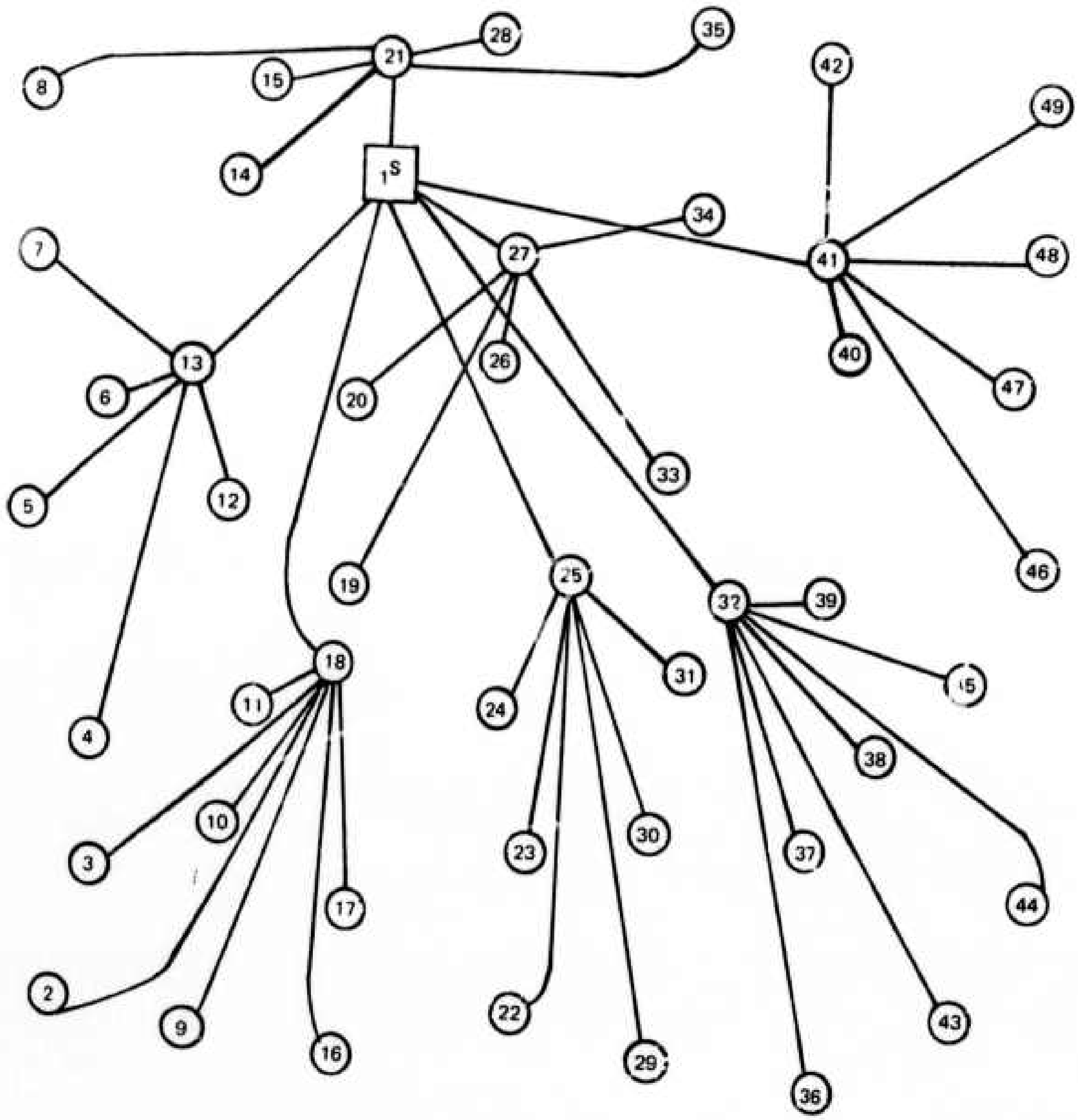


Figure 14.4: CCSDR (HI) System With Improved Labeling

the summary of other measures is given in Table 14.2. The comparison demonstrates that the CCTRD system is superior to the CCSDR system, in terms of throughput, delay, and other measures. One can see that the CCSDR system is saturated at an offered rate of about 13%.

14.2.1 Effect on Blocking Level

In Table 14.2, one can see that one reason for the relatively low throughput of the CCSDR system at an offered rate of 25% is due to blocking. Furthermore, the fraction of time

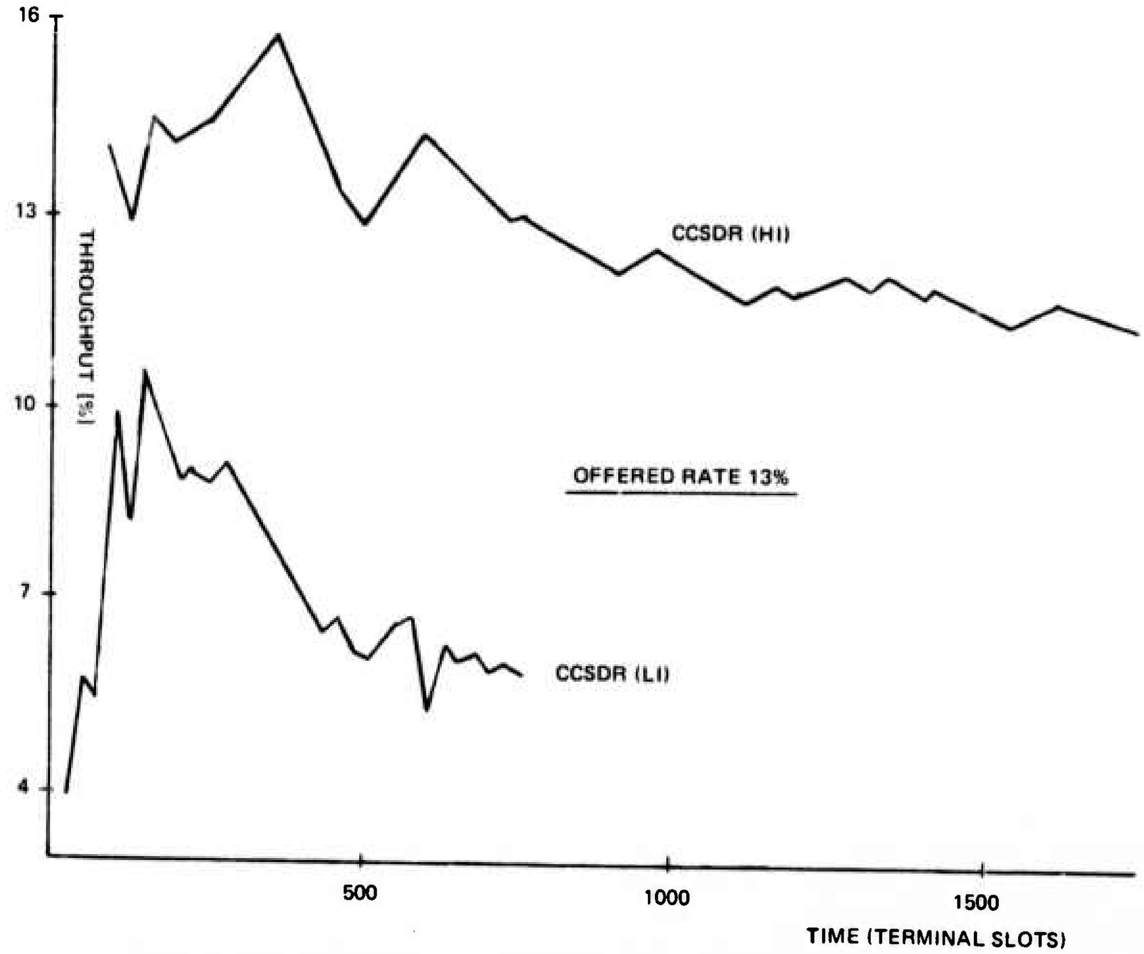


Figure 14.5: Throughput vs. Terminal Slots: CCSDR (HI) and CCSDR (LI)

Table 14.1: Effect of Range On Network Performance For Single Data Rate System

Interference Level	Offered Rate (%)	Throughput (%)	Delay Of IP To Station (Terminal Slots)	Rate Of Station Response	Prob. Station Busy	% Of IP Blocked	Total % Of IP Loss	Terminals Remaining
CCSDR (LI)	13	5.95	40.11	1.14	.53	2.98	32.83	13
CCSDR (HI)	13	10.55	23.93	1.81	.43	9.83	9.83	13
CCSDR (HI) (Improved Labels)	13	12.14	16.61	2.06	.50	10.63	11.41	10

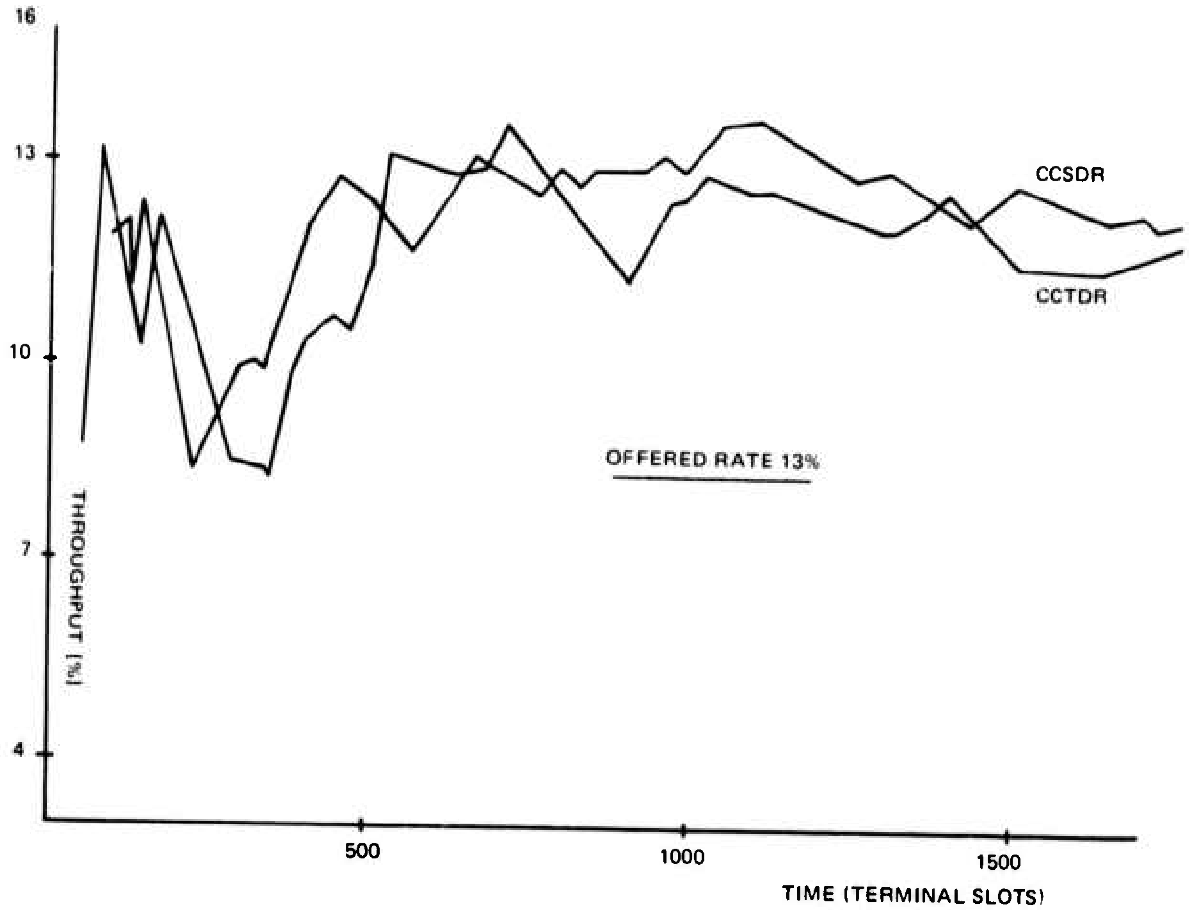


Figure 14.6: Throughput vs. Terminal Slots: 13% Rate

that the station is busy has decreased. This may suggest that the station may be able to handle more terminals providing they are able to enter the system. To examine this point, we ran the CCSDR system with an offered rate of 25%, and relaxed the constraint for entering the system. Rather than resulting in better performance, this step resulted in reduction in blocking and increase in delay. The throughput increased to 12.63%, the blocking decreased to 18.35% and the total loss decreased to 30.73%. On the other hand, the delay increased to 57.82, the fraction of time the station is busy increased to .57, and the rate of station response decreased to 1.32.

To conclude, when we enabled more terminals to enter the system, the throughput increased insignificantly, from 12.20% to 12.63%; on the other hand, the average packet delay *increased significantly*, from 34.97 to 57.82 terminal slots. This suggests that one of the important design problems in the packet radio system is the blocking level of terminals.

14.3 Throughput, Loss, and Delay of CCSDR and CCTDR Systems

Similar to curves of throughput versus channel traffic, for which the relation is known analytically [38], we can attempt to draw curves of system throughput vs. offered rate for

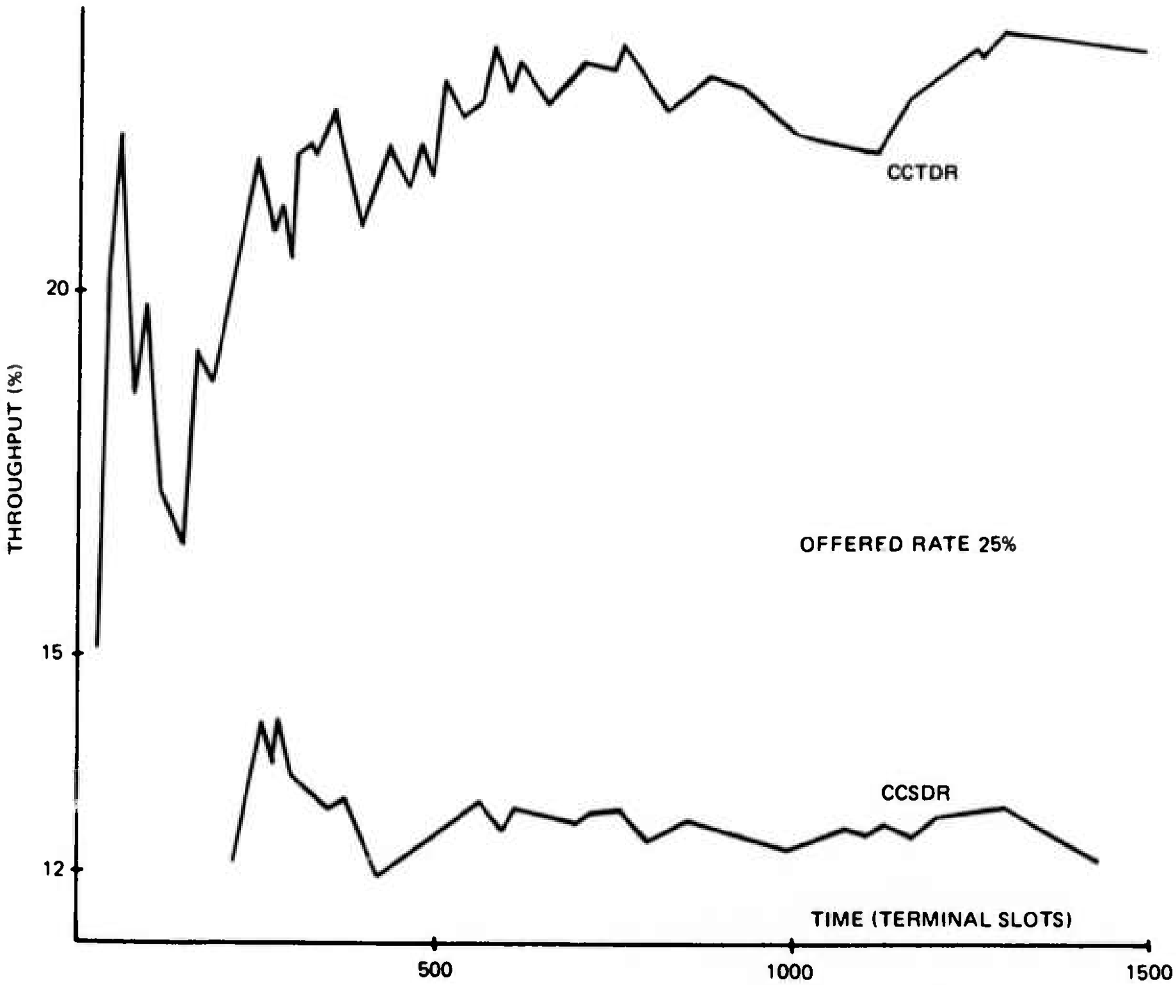


Figure 14.7: Throughput vs. Terminal Slots: 25% Rate

Table 14.2: Single Data Rate vs. Dual Data Rate System Performance

	Offered Rate (%)	Throughput (%)	Delay Of IP To Station (Terminal Slots)	Rate Of Station Response	Prob. Station Busy	% Of IP Blocked	Total % Of IP Loss	Terminals Remaining
CCSDR	13	12.14	16.61	2.06	.50	10.63	11.41	10
	25	12.20	34.97	1.61	.48	29.50	32.95	23
CCTDR	13	12.39	4.91	1.99	.26	1.59	1.59	9
	25	23.33	11.51	1.97	.31	3.31	3.31	34

estimating the maximum throughput. Figure 14.8 shows the throughput versus offered rate for CCSDR and CCTDR systems. The curves are linear for low offered rates and saturate when the offered rate increases.

For the CCSDR system one can see that the throughput is practically the same when the offered rate is increased from 13% to 25%. This and the other measures (see Table 14.2), (for example, the rate of station response) show that the system is overloaded at a 25% offered rate. On the other hand, the system seems to operate at steady state at an offered rate of 13% (rate of station response 2.06). A rough estimate of maximum throughput for this system would be between 12% and 15%. Similar observations of the performance measures lead to an "estimate" of between 27% and 30% for the maximum throughput of the CCTDR system in the specified repeater configuration.

The average delay of the first Information Packet from terminal to station, and the Total Loss, as a function of offered rate are shown in Figure 14.9 and Figure 14.10 respectively.

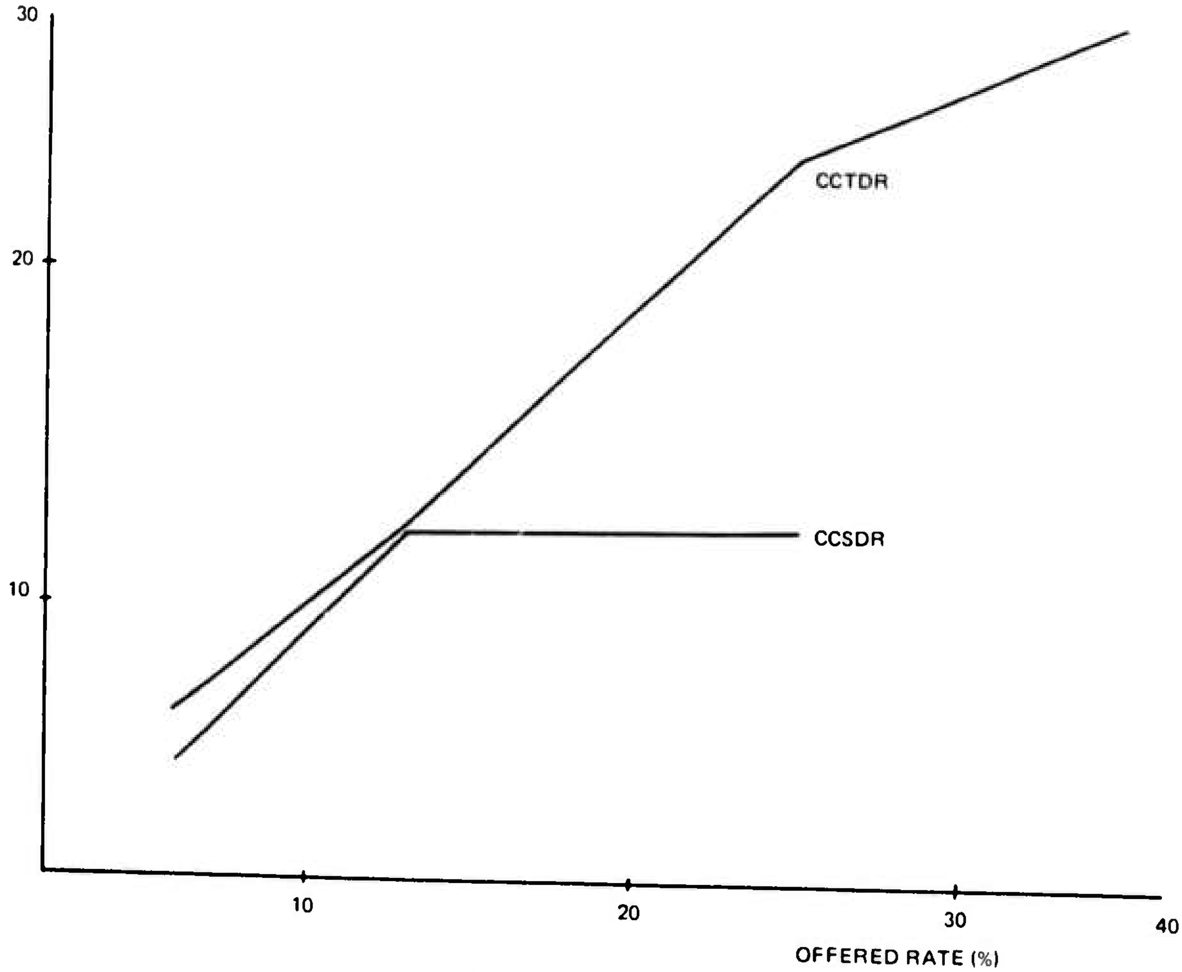


Figure 14.8: System Throughput vs. Offered Rate

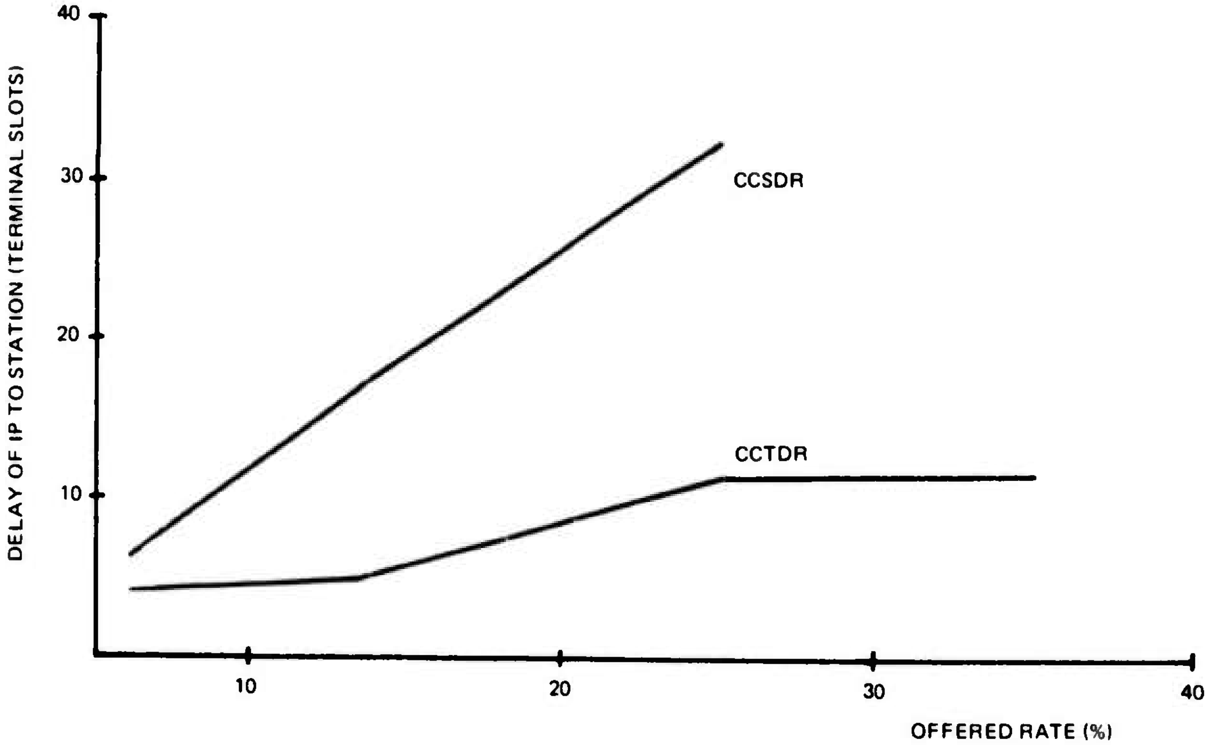


Figure 14.9: Terminal-Station Delay vs. Offered Rate

REMARKS: There are many parameters in the simulation program which we have not experimented with (or tried to optimize) and which affect the quantities discussed above. One parameter which is significant in determining the maximum throughput is the average number of response packets from station to terminal. The effect of this parameter has been analyzed in [28], for a slotted ALOHA random access mode. It has been shown that the maximum throughput is increased in the Common Channel system when the rate of response increases, and the maximum throughput tends to 100% of the data rate when the rate of response tends to infinity. We expect that this parameter has a similar effect for the mode of access simulated. In the results reported here the rate of response is 2.0 which is small compared to usual estimates for terminals interacting with computers. Furthermore, the relatively short terminal interaction increases the traffic overhead of the search procedure per information packet.

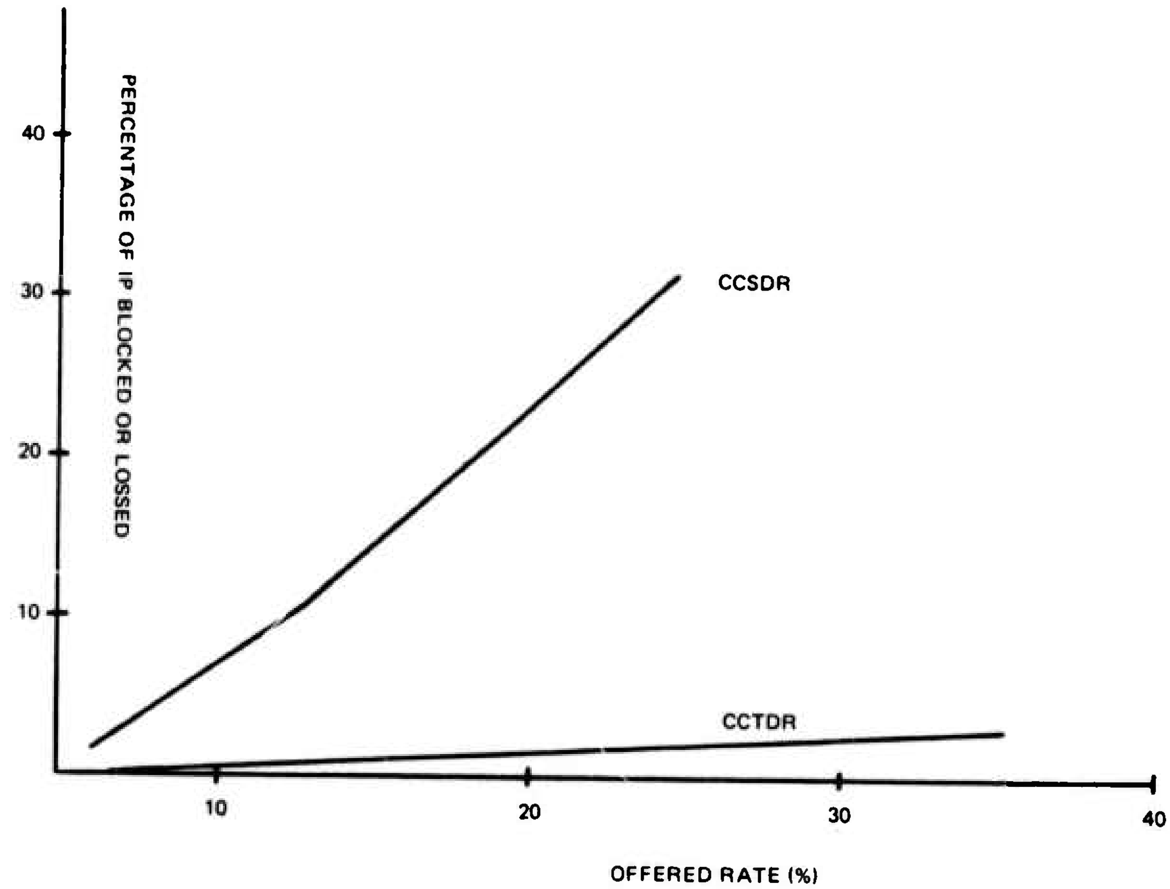


Figure 14.10: IP Blocking vs. Offered Rate

Chapter 15
PACKET RADIO SYSTEM AREAS FOR FURTHER STUDY

As is evident, packet radio offers a new and challenging area for network analysis and design. Previous studies have merely touched crucial areas. Further study to develop methodology, to provide support for hardware and software design, and to effectively control and manage network resources is required. A variety of studies is currently underway. These studies will:

a. Estimate system capacity as a function of terminal-repeater and repeater-repeater signaling rates for multistation networks.

b. Compare the performance of systems with varying degrees of receiver capture, multiple channels, directional antennas and multiple detectors.

c. Determine efficient operating parameters including time out intervals, handover numbers, and number of retransmissions.

d. Determine relationship between number of repeaters, throughput, delay and blocking.

e. Compare the efficiency of direct terminal to terminal routing versus hierarchical routing in multistation networks.

f. Estimate throughput, delay, and blocking for multistation networks.

g. Develop and test multistation algorithms for routing and labeling and relabeling.

h. Develop high level global flow control algorithms to allow effective utilization of system resources.

i. Determine network control strategies to identify and monitor network congestion and element failure conditions.

j. Formulate dynamic reliability and survivability criteria and develop algorithms for network reliability analysis and design.

k. Develop algorithms for configuring packet radio networks to meet specified reliability and survivability criteria.

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